THE WORLD'S BEST RECORDING TECHNOLOGY MAGAZINE — 1985 – 2024

SOUND ON SOUND™

MUSIC PRODUCTION TECHNIQUES / INDEPENDENT IN-DEPTH PRODUCT TESTS / ENGINEER & PRODUCER INTERVIEWS / LIVE SOUND

ASTROLAB

ARTURIA'S GREATEST HITS — IN ONE KEYBOARD!

Neve 1073SPX-D

Classic preamp with USB interfacing

WIN

SSL ANALOGUE MASTERING CHAIN

WORTH $5394

Oeksound Bloom

A 'magic mirror' for audio?

FNZ

The producers who brought back sampling

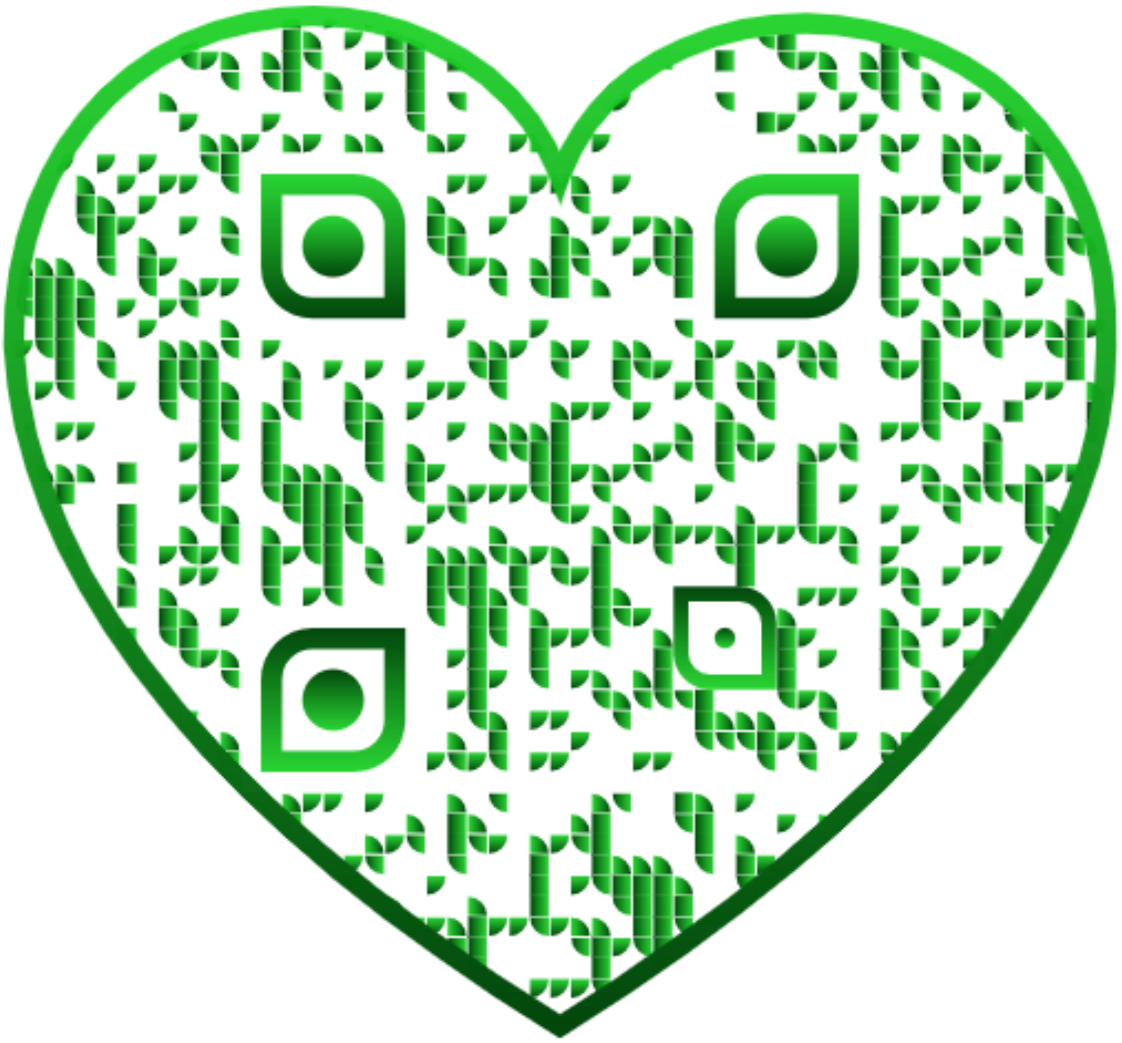
ON TEST: MINIMAL AUDIO / EARTHWORKS / IK / SPL / AUDIO MODELING / WARM AUDIO / LEWITT / TONE PROJECTS / REMIC / REMUSE

TECHNIQUE: MIX RESCUE / VU METERS / DAW WORKSHOPS

www.soundonsound.com

0 29074 05361 3 05

USA $11.99 / Canada C$12.75



NEW: **SWAM STRING SECTIONS!**

SWAM engine

Audio Modeling

"Wow! I love the sound, rich and warm. And just as agile as Audio Modeling's other products. The dynamic range is the most I've heard from any library and the quick access to tons of articulations makes it very enjoyable to play!"

Guy Moon, Award-Winning Film and TV Composer

www.ilio.com | 800.747.4546

ILIO

GOING THE DISTANCE

One of the good things about analogue gear, supposedly, is that it doesn't suffer from built-in obsolescence. Software licences become useless if the manufacturer fails to keep their product current. Digital hardware often depends on driver updates to keep it operational. But as long as there are patchbays, there'll always be a home for our analogue compressor or EQ. And in a world where sustainability is a pressing concern, we should all aim to own things that are timeless rather than disposable.

In practice, things are rarely as simple as that. For one thing, there's plenty of ancient digital gear that still works perfectly well. Your Publison delay or PPG synth won't become a doorstop just because those companies aren't issuing new firmware. The life-limiting factor with older digital kit is servicing and the availability of parts, just as it is with most analogue equipment.

Even with contemporary equipment, though, the risks of built-in obsolescence are often overstated. Yes, there have been some shocking and high-profile cases of abandonware, but when audio interfaces go to the great driver architect in the sky, it's often not because of compatibility issues. It's because they have become physically unreliable, or superseded by better-sounding, more powerful replacements. And wasn't that exactly what happened with analogue mixing consoles, back in the day?

Large-format consoles and other classic gear were, at least, designed to be easily repaired and maintained, and that certainly isn't true of modern digital hardware. But neither was it ever true of analogue circuits encapsulated in goop, or integrated circuits with identifying markings scratched off. And it's not only digital gear nowadays that uses surface-mount components and doesn't come with schematics.

Finally, although analogue gear may be durable, its usefulness in the modern studio often depends on digital tools that allow us to create an environment in which it can be integrated. And the flip side of that is that when those digital tools aren't available, even being fully analogue does not offer unlimited protection against obsolescence.

For example, most Ambisonic microphones generate an entirely conventional four-channel analogue signal. Before it reaches the listener, this output needs to be matrixed to B-format and then decoded to a virtual mic or speaker array. In the more upmarket Soundfield models, this is done in hardware, but most Ambisonic mics rely on software processing. Yet, at the time of writing, there are no Apple Silicon native plug-ins that can convert A-format to B-format; and I know of only one developer, Blue Ripple, offering compatible Apple Silicon plug-ins for processing the B-format signal.

For now, Ambisonic mic owners can run legacy code under Rosetta — but what will become of our precious all-analogue devices when that is no longer possible?

Sam Inglis
Editor In Chief

"In a world where sustainability is a pressing concern, we should all aim to own things that are timeless rather than disposable."

SOUND ON SOUND

SOUND ON SOUND LTD (HEAD OFFICE)
ALLIA BUSINESS CENTRE
KING'S HEDGES ROAD
CAMBRIDGE, CB4 2HY, UK
T +44 (0)1223 851658
sos@soundonsound.com
www.soundonsound.com

WWW.SOUNDONSOUND.COM/SUBSCRIBE

WORLDWIDE EDITIONS

NORTH AMERICA

UK/WORLD

EDITORIAL
sos.feedback@soundonsound.com

Editorial Director Dave Lockwood
Executive Editor Paul White
Editor In Chief Sam Inglis
Technical Editor Hugh Robjohns
Reviews Editor David Glasper
Reviews Editor Matt Houghton
Reviews Editor Chris Korff
Production Editor Chris Korff
News Editor Luke Wood

SUBSCRIPTIONS
subscribe@soundonsound.com
www.soundonsound.com/subscribe

Circulation Manager Luci Harper
Administrator Nathalie Balzano

PRODUCTION
graphics@soundonsound.com

Production Manager Michael Groves
Designer Alan Edwards
Designer Andy Baldwin
Designer Mick Reilly

ADMINISTRATION
admin@soundonsound.com

Managing Director/Chairman Ian Gilby
Editorial Director Dave Lockwood
Sales Director Robert Cottee
Marketing Director Paul Gilby
Finance Manager Keith Werthmann

ONLINE
support@soundonsound.com

Digital Media Director Paul Gilby
Design Andy Baldwin
Web Content Editor Callum Hall
Web Editor Adam Bull
Podcast Production Manager Atheen Spencer
www.soundonsound.com
twitter.com/soundonsoundmag
facebook.com/soundonsoundmag
instagram.com/soundonsoundmag

DISTRIBUTION
distribution@soundonsound.com

International Distribution
Magazine Heaven Direct
www.magazineheavendirect.com

ADVERTISING
usadsales@soundonsound.com

Sales Director Robert Cottee
North America Sales Manager Dan Brown
Regional Sales Manager David Carson
UK Media Sales Manager Guy Meredith

MARKETING
marketing@soundonsound.com

International Business Development
Nick Humbert

Printed in the USA
Not for re-sale outside North America

ISSN 1473-5326
A Member of the SOS Publications Group

The contents of this publication are subject to worldwide copyright protection and reproduction in whole or part, whether mechanical or electronic, is expressly forbidden without the prior written consent of the Publisher. Great care is taken to ensure accuracy in the preparation of this publication but neither Sound On Sound Limited nor the Editor can be held responsible for its contents. The views expressed are those of the contributors and not necessarily those of the Publisher or Editor. The Publisher accepts no responsibility for the return of unsolicited manuscripts, photographs, or artwork.

© Copyright 2024 Sound On Sound Limited. Incorporating Music Software magazine, Recording Musician magazine, Sound On Stage magazine, SPL magazine, Sound Pro magazine and Performing Musician magazine. All rights reserved.

All prices include VAT unless otherwise stated. SOS recognises all trademarks.

MASSIVE SOUND

NOW WITH
PROPHET VS
WAVETABLES!

24 voices · 3 osc per voice

both PPG-lineage and modern wavetables · built-in wavemaker

GROOVE SYNTHESIS
groovesynthesis.com

NOW IN TWO SIZES

NEW!
- FREE-RUNNING OSCS
- POLY UNISON
- LO-FI WAVETABLE IMPORT

· 4 multi-parts · analog & digital filters

· virtual analog oscs · sequencer · 2 FX per part · 4 LFOs · 6 envelopes

3RD WAVE
ADVANCED WAVETABLE SYNTH

IN THIS ISSUE

www.soundonsound.com

May 2024 / issue 7 / **volume 39**

FEATURES

92 **Modular**

We talk integrated reverb and take a quick look at what's new in the world of modular.

94 **Mix Rescue**

The foundation of a good, engaging mix is a strong arrangement. But how far can you go to improve things at the mixing stage?

102 **VU Meters: Virtually Useless Or Very Useful?**

After 85 years of active service, the humble VU meter remains as useful as ever in today's digital studios — even though BBC engineers nicknamed it 'virtually useless'!

120 **Spotlight: Workstation Synthesizers**

Looking for an all-in-one playing and sequencing solution? Look no further!

124 **Alan Moulder: Why Mentors Matter**

If there's one thing that engineers and producers need above all, it's a good mentor and role model — and they don't come much better than MPG Icon Alan Moulder.

128 **Korg MS Series Revisited**

Korg's MS synthesizer range contains some *bona fide* classics — and is much more extensive than you might imagine.

136 **Inside Track: FNZ (Finatik & Zac De Boni)**

Hard work and a love of sampling have made FNZ the hottest production duo around.

146 **Why I Love... The Suzuki Omnichord**

Ben Brockett on how an unexpected gift ignited his love affair with the Suzuki Omnichord.

50 NEVE 1073SPX-D

ON TEST

70 **Amphion One25A**
Active Monitors

COVER 44 **Arturia AstroLab**
Performance Keyboard

36 **Audio Modeling SWAM String Sections**
Modelled Orchestral Strings Instrument

12 **Earthworks SR117**
Vocal Microphone

86 **Electro-Harmonix Pico Triboro Bridge**
Overdrive Pedal

64 **Gauge ECM-87 Virtual Mic Locker Kit**
Microphone Modelling System

84 **Hit'n'Mix RipX DAW PRO**
Source Separation & Audio Processing Software

54 **Hologram Electronics Chroma Console**
Multi-effects Pedal

78 **IK Multimedia ARC Studio**
Speaker Correction System

20 **Imaginando BAM Beat Maker & Music Maker**
Music Production Software

90 **Knob Technology SGR1806-20**
Eurorack Module

16 **Lewitt RAY**
Capacitor Microphone With Distance Sensing

58 **Minimal Audio Current**
Software Synthesizer

14 **MountainRoad DSP Lumina Delay**
Delay Plug-in

8 **Neural DSP Morgan Amps Suite**
Guitar Rig Modelling Plug-in

50 **Neve 1073SPX-D**
Channel Strip & USB Audio Interface

28 **Oeksound Bloom**
Adaptive Tonal Shaping Plug-in

91 **Qu-bit Electronix Mojave**
Eurorack Module

66 **Remic Reshape**
Instrument Microphones

24 **ReMuse ReMuse:KIT**
Drum Separation Software

86 **Rode NT1 Signature**
Cardioid Capacitor Microphone

142 **Sample Libraries**
Emergence Audio Viola Textures
Spitfire Audio Crystal Bowls By Aska Matsumiya
Best Service Horizon Leads By Sonuscore
Cradle State Machine Slow Drift

74 **SPL Channel One Mk3**
Channel Strip

40 **Tone Projects Hendyamps Michelangelo**
Analogue-modelling EQ Plug-in

88 **Universal Audio UAFX Brigade**
Chorus & Vibrato Pedal

10 **Warm Audio RingerBringer**
Ring Modulator Pedal

82 **Westwood Instruments Lost Synth**
Software Synthesizer

90 **Xaoc Devices Ostrawa & Bohumin**
Eurorack Module

WORKSHOPS

108 **Digital Performer**
112 **Studio One**
114 **Pro Tools**
116 **Logic**
118 **Cubase**

Neural DSP
Morgan Amps Suite

JOHN WALDEN

If you were to ask me to name one fantasy guitar amp that I'd love to own but could never justify the cost, I'd say "a Morgan SW22R". Morgan Amps definitely fit the 'boutique' label: they're a relatively small company making high-end, hand-crafted products, many of which are inspired by classic amp designs. So, for example, Morgan's AC20 is their take on the Vox AC30, the PR12 is inspired by Fender's iconic Princeton Reverb, while the SW22R and SW50R have their roots in the (also boutique) Dumble. All are no-expense-spared recreations of long-standing original designs, and built to the highest standards. And now, thanks to a collaboration between Morgan Amps and those clever brains at Neural DSP, I can afford access to these sounds through Neural's latest plug-in, the Morgan Amps Suite, which emulates the Morgan SW50R as well as their slightly more wallet-friendly AC20 and PR12.

Guitar Rig Modelling Plug-in

If you're looking for a plug-in that beautifully recreates the classiest of boutique amps, read on...

Amps, Cabs & Effects

Following the usual Neural DSP format, you get not only the three amps but also a compact collection of virtual stompboxes, including an excellent tremolo, a flexible cab simulation based on a Morgan 1x12 Celestion-equipped cabinet with dual mic configurations, an IR-loader, a nine-band EQ, a studio-style delay (tape-based in this case), and reverb. The GUI includes an easy-to-use preset browser, transpose options, a very effective Doubler for an instant double-tracked effect, a tuner, and a metronome. It can run standalone or as a VST, AU or AAX plug-in in Mac and Windows hosts. But while all these extras are excellent, the amps are definitely the headline act here!

True to the Morgan hardware, all the models provide single-channel amps with fairly compact control sets. So, for example, on the AC20, the Vol knob controls the preamp gain, and the Power knob the power-amp level, while the Cut knob is a tone control (the equivalent of the AC30's Tone Cut). As on the actual AC20, the Bright and Bass Cut switches provide additional tonal options and, for total accuracy (if perhaps not essential in an amp sim), there are also modelled Standby and Power switches. The PR12 and SW50R control sets are equally streamlined, but both also recreate the built-in reverbs those amps feature.

What's really impressive about all three amp models is their realistic response. These are perhaps all best described as 'clean' amps and, providing you get your guitar DI input levels right (starting at zero, dial in as little gain on your input as you can get away with), you can replicate that with all three models. However, as you push the amp controls further, things will start to break up in a really satisfying fashion, with a character that's true to the original hardware.

If you want to go further, then the stompbox pedals provide plenty of additional gain possibilities. All flavours of classic blues and rock can be found easily, and there are metal tones if you push the more aggressive of the two overdrive stompboxes. The preset collection demonstrates this range very ably too — whether you want Chris Isaak cleans (the Blue Hotel preset), super classy funk (Strat 4th Position Funk; the compressor pedal is very good), John Mayer's singing strat (Slow Dancing Bell Tone), Brian May's crunch (Red Special Rhythm), or something beyond, there are good presets to get you started.

Simply The Best?

Neural DSP really are very good at what they do. In fact, as long as you're happy to dip into their catalogue for the specific titles that suit your needs, I'm not sure that there's a better way to access authentic software recreations of specific amps — if you're looking to add some truly classy clean-to-crunch options to your 'guitarsenal', the Morgan Amps Suite plug-in is currently as good as it gets. ■■■

summary

One of the best amp emulations out there, this plug-in delivers the sound of three of Morgan's most desirable boutique amps for a fraction of the price. Fabulous stuff!

$ €99 (about $100).
W www.neuraldsp.com

m908. Immersive Monitoring, Perfected.

The m908 monitor controller gets you working in any format quickly and easily, expertly managing any speaker system from stereo to Dolby™ Atmos 9.1.6. Firmware 2.0 includes our new web-based control platform which lets you operate and configure the entire system from any web browser (desktop or mobile). Additionally, the m908's room correction EQ capability provides 12 bands on all 24 channels at all sample rates. With unrivaled audio performance and mechanical elegance, the m908 is simply the finest all-in-one monitoring tool for modern music production.

"Grace is renown for incredible sonics, however the m908 is an entirely different machine. Not only does it sound phenomenal, but the functionality and routing is on another level. As the centerpiece of our Dolby Atmos room, it handles hundreds of I/O, multiple speaker sets, room correction, and so much more. I can't say enough great things about it! "

Glenn A Tabor III
Multiple Grammy™ winning producer / engineer, Gat3 Studio (www.gat3.com)

GRACE DESIGN
www.gracedesign.com

Warm Audio RingerBringer

Ring Modulator Pedal

Can this ring modulator live up to the standards of the old MoogerFooger?

PAUL WHITE

Available for about half the price of a second-hand MoogerFooger Ring Modulator pedal, the all-analogue Warm Audio RingerBringer purports to be a faithful, true-to-spec recreation of Moog's revered original. It even shares the same cosmetic vibe, complete with wooden end cheeks, and we're told that all the ICs and transistors are hand-selected for optimum performance.

Ring Tones

Ring modulators are something of an acquired taste and tend to appeal to those who like to experiment: their results can sound somewhat dissonant, but they do offer an easy way to explore less conventional sounds. The reason a ring modulator sounds so odd is that it combines two signals such that their sum and difference frequencies are sent to the output, but with none of the original frequencies present. This often means that the output bears no 'musical' relationship to the input, unless the modulation frequency is set very low. However, by adding a wet/dry mix control, it's possible to mix in just enough of the ring-modulated sound to flavour the original source without overwhelming it.

As with most ring modulator pedals, the RingerBringer has an internal oscillator used to provide the carrier signal, and this is switchable between two frequency ranges that cover very slow modulation right up to and beyond audio-frequency modulation (0.6Hz to 80kHz). Feed in a voice with a carrier set between 50 and 100 Hz and you'll hear the familiar Dalek voice.

This pedal has two sections: LFO and modulator. The modulator section has controls for Mix and Frequency along with a Lo/Hi frequency range switch. The LFO (0.1 to 25 Hz) doesn't modulate the audio directly as it might in, for example, a tremolo pedal but instead modulates the frequency of the carrier oscillator. This has controls for Amount and Rate, along with a switch to select between sine or square wave. Between the two sections is a drive control, the main function of which is to allow weaker signals to be brought up to a practical working level, though when pushed fully clockwise it will also add some harmonic distortion. Note that the drive control is always active even then the pedal is bypassed.

Status LEDs indicate input level, LFO speed and bypass, the latter being controlled by a conventional footswitch. Power can come from a battery or an optional 9V PSU. As the current draw is 100mA, using a PSU is perhaps more practical.

Look around the back of the pedal and you'll see exactly the same appointments as on Moog's original, and that should make the pedal particularly attractive to modular synth users. In addition to audio in and out, there's a row of four jacks that can accept control voltage or expression pedals to control Rate, Amount, Mix and Frequency. There are also additional output jacks for the LFO and carrier signals as well as a carrier input jack that allows an external audio signal to replace the internal carrier oscillator.

Like the MoogerFooger that inspired it, the RingerBringer has a wealth of rear-panel connections that should appeal to modular synth lovers.

Ring Of Truth?

I have to confess that it's been a couple of years since I spent much time with a Moog ring modulator, but to my ears this one produces the same subjective results — it can certainly sound decent, and it spans an enormous range. At low modulation speeds, the pedal conjures up a pleasing tremolo effect, with a hint of vibrato thrown in, while at higher settings it goes from gargling and growly, right up to shrieking mayhem. Using the LFO to modulate the carrier also brings in some welcome movement, even at very slow modulation frequencies. Guitarists will find some useful effects providing they are used sparingly, but I suspect that it's modular synth users who will get the most out of the RingerBringer, because of those rear-panel CV connections. They are also in a better position to experiment using source sounds (and carrier sounds, come to that) with different waveforms. If you are in the market for a used original but can't afford one, I think you'll be very happy with the RingerBringer.

summary

A good-sounding ring modulator with plenty of range.

$ $219.
W https://warmaudio.com

The 2448 Console.
To get a personalized online demo of the 2448, 1608-II or THE BOX Console
schedule your **Virtual Console Experience** at
apiaudio.com/DEMO
api

Earthworks SR117

Vocal Microphone

Earthworks' latest stage mic punches way above its price tag!

PAUL WHITE

Earthworks built their reputation on very accurate, small-diaphragm studio capacitor microphones. They've since branched out into broadcast, podcasting and live circles, with the SR117 being their latest and most affordable stage vocal microphone. Outwardly, it looks like a typical dynamic stage mic, but it too is a capacitor model. Like most live mics, the SR117 has a foam lining in the basket that can be removed for cleaning, but here there's also a removable internal fine-mesh cylinder covering the capsule, which aids resistance to popping. The body, which is finished in black with a stainless steel ring just below the basket, has the familiar tapered shape with the XLR at the bottom end, and the mic feels reassuringly solid, weighing in at 380g. There's no switch, which is fine by me, as vocalists always seem to mess with them! A soft case and a stand clip are included.

The capsule, which can be seen clearly after removing the basket and internal mesh screen, has an overall diameter of around 10mm and is quite rigidly supported, though handling noise didn't seem to be at all problematic. The necessary porting to produce the mic's supercardioid response is built into the capsule, which has a tall, narrow profile.

The polar pattern is remarkably consistent across the frequency range, and the mic's response is virtually dead flat from 20Hz upwards other than a slight bump at 10kHz, before falling to -10dB at 20kHz. Having a consistent polar pattern no doubt aids resistance to feedback, as well as minimising tonal changes when the singer moves off-axis, though some tonal changes at varying mic distances are inevitable due to the proximity effect. Because there's no built-in presence peak, the mic can be tailored to any voice type with EQ.

Being a capacitor microphone, phantom power (48V) is needed for operation, but the payoff is that you get the performance of a studio capacitor microphone, including a sensitivity of 5mV/Pa. A peak SPL handling of 140dB is quoted. The signal-to-noise ratio is 74dB, equating to a self-noise of 20dB SPL A-weighted. While this is not a particularly low noise figure for a general-purpose studio microphone, in the intended application of close-miking vocals, it is more than adequate. Having said that, the mic can also be used with instruments, either live or for recording, where its high SPL handling enables it to cope with the likes of drums and horns as well as more gentle sources such as acoustic guitar. Indeed, the SR117 is eminently suitable for studio as well as stage use, so it can easily do double duty for home studio owners who also play live.

Quiet Riot

On first listening the SR117 may seem less airy or 'forward'-sounding than some of the more common vocal mics, but I think this is a strength rather than a weakness, because mics with a distinct character usually work better with some voices and less well with others. If you need more presence or air, a hint of EQ will deliver it. Having said that, I used the SR117 with the desk EQ set flat for a live performance with a female vocalist and it sounded perfectly balanced just as it was. It really did have the open, natural sound of a good studio mic! Resistance to feedback was solid, handling noise very low and there was plenty of level, so I didn't need too much by way of mixer preamp gain. With some vocalists you may need a low-cut filter to avoid popping, given the extended low-end response of this mic, but the internal pop screen seems to make a big difference, and I didn't hear any mic popping at all during my live tests.

Leaving the best until last, this high level of performance comes at a surprisingly low price. While other Earthworks live vocal mics retail at several hundred dollars, the SR117 is available for well under $200. Given the quality of engineering and of course the sound quality, that represents exceptional value, putting the SR117 within easy reach of semi-professional performers. I expect to be seeing many more of these mics on stages, in clubs and in pubs before long.

summary

A versatile live vocal mic that can also hold its own in the studio. Does double duty with instruments, too.

$ $199
W www.earthworksaudio.com

NOT A CLONE IN SIGHT

The world's first
harmonic eq®

INNOVATE YOUR SOUND

CRANBORNE AUDIO

www.cranborne-audio.com

MATT HOUGHTON

MountainRoad DSP Lumina Delay

Delay Plug-in

Tired of the same old delay effects? With Lumina, you can draw, position and shape your repeats any way you like.

Available for Mac and Windows and supporting AAX, AU and VST3 hosts, Lumina Delay caught my attention because it's different from the countless other delays in my plug-in folder: rather than setting familiar parameters such as the delay time and feedback level with knobs or faders, you click to place each repeat on a sequencer-style grid. Download and installation was quick and easy: on purchasing, you receive an email with a licence code, and on first loading you just enter your email and that code. Job done.

Overview

On the main grid you can set the level of each repeat, position it in the stereo panorama, and shape it with high- and low-pass filters, with each parameter given its own row. To 'paint in' a pattern of repeats, click on the grid and a repeat will appear, with controls in each row for manipulating it. This grid extends to eight bars, and you can set the resolution from whole bars down to 64ths, and specify normal, dotted or triplet. At the top, a red bar resembling a DAW's playback selection loop allows you to zoom in/out to accommodate everything, or focus right in on the details (you can also do this with modifier keys and scrolling). The grid also adapts to your DAW's time signature automatically, which is nice.

On each row is a simple, intuitive control to adjust its parameter. For level, it's a dot that you drag up or down to anything from minus infinity (muting but not deleting the repeat can be handy when experimenting) to +24dB, the middle position being unity. You can drag this dot to change the timing. The stereo pan control is similar, but with the centre-panned position in the middle, left at the top and right at the bottom. As of v1.2 each dot for level and pan lights blue when the repeat plays and red to indicate clipping — a neat touch. The filters share a row whose control is a vertical bar, the top and bottom edges of which can be click-dragged to adjust the high- and low-pass filter frequencies, while moving the whole bar up/down tweaks both.

Global controls allow you to adjust each parameter separately, for all repeats, and the changes are reflected on the grid. There is also a global 'delete all taps' button, and some handy shortcuts. For instance, there are modifier keys to click and delete a node, to increase precision while dragging, and to duplicate a repeat. You can also right-click to bring up a 'precision' menu, where you can reset the node, specify the time in milliseconds, and edit the parameter that node controls — very useful when the grid starts to get crowded.

In Use

There are presets to help you get started and these can be fun, but Lumina is so simple to use that you won't need them: you can create any delay pattern you desire, with whatever panning and filtering you want on each repeat, with shockingly little effort. It's even possible to draw in simultaneous repeats that have different characteristics — if you want a low-passed sound in the left speaker and a high-passed one on the right, you can do that.

Since the repeats don't automatically get quieter or degrade as they might with your usual delay, you really can design some incredibly complex rhythms and textures that you just couldn't with a typical delay. You could have a series of delays getting louder or brighter over time in one speaker and quieter/darker in the other, for example. Or have alternate repeats or every third repeat brighter or darker, softer or more strident than the others. Or automate the bypass to bring in machine-gun stuttering as a special effect. The only limit is your imagination.

Given how Lumina works, there's no feedback control (that would get very messy very fast), and there's no single knob you can turn for more complexity or a longer tail. The presets are all set to 50 percent wet too (a 'wet lock' facility to prevent presets changing the setting would be useful). I had very interesting discussions with MountainRoad on these points and more, and we should see some significant developments soon, including some more exciting ones about which I can't divulge details now. But I think I *can* hint that there will be additional parameters available to process the repeats!

Already, though, Lumina is incredibly useful, it does something no other delay I have can do and it does it with minimal fuss. Sound designers should love it but it absolutely has a place in mixing, so if you've been pining for more control over your delays, go check out the demo!

summary

With its novel approach, Lumina already reaches the parts other delays cannot reach — and it's only going to get better from here!

$ $149 (discounted to $99 when going to press).
W https://mountainroaddsp.com

CQ

MIX SMARTER WORK FASTER SOUND BETTER

Sound your best on stage, in rehearsals, and in the studio.

CQ makes it easy, whatever your mixing style. Musicians can jump in with instrument-specific Quick Channels and dynamic FX, tweak their monitor mix via an app and let our intelligent Gain and Feedback Assistants handle the details.

Engineers will love the in-depth channel processing and apps for remote position mixing. And multitrack recording couldn't be easier with an integrated USB interface and SD Card slot.

WINNER
SOS AWARDS 2024

FIND OUT MORE AT ALLEN-HEATH.COM/CQ

ALLEN&HEATH

Lewitt RAY

Capacitor Microphone With Distance Sensing

Have Lewitt invented the cure for poor mic technique?

SAM INGLIS

Phantom power was designed for solid-state capacitor microphones, and its limitations reflect that. Although the mic's capsule needs to be polarised, this doesn't really draw any current, so the only thing that's being powered as such is the impedance converter, which typically contains just a couple of active components. The 10mA maximum phantom power current draw is enough for these applications, but it's a limit that you run up against pretty quickly if you want to power other active circuits. Nevertheless, enterprising designers have done creative things with the meagre resources available.

Scope Labs' Periscope mic, for example, incorporates a phantom-powered analogue compressor, while the UA Sphere mic has a built-in oscillator to calibrate your mic preamp input level, and LED indication of switch and button settings is now almost commonplace. But Lewitt's new RAY microphone takes the idea of built-in, phantom-powered processing to several new levels.

In Black & White

In many ways, the RAY can be thought of as an evolution of Lewitt's existing LCT 440 Pure. Like that product, it's a large-diaphragm true capacitor microphone with a one-inch, centre-terminated capsule and a fixed cardioid polar pattern. The RAY has the same form factor as the 440 and ships with the same accessories, including an effective shockmount, a magnetically attached pop shield and a foam windshield. The two mics also have the same form factor, with an attractive rectangular shell and a very open headbasket.

So what's special about the RAY? Well, once you've realised that the side with the large Lewitt logo on is actually the back of the mic (gets me every time), you'll notice that the front side is adorned with something resembling the Abbey Road zebra crossing logo, plus two buttons labelled Aura and Mute. Closer examination will also reveal a pair of racetrack-shaped proximity sensors located either side of the black-and-white steps. The RAY uses these to detect how far away the performer is — and modify its response appropriately.

RAY Tracing

I'll say that again, because it is really quite a novel idea: the RAY is a mic that can follow a performer's movements in real time, and adjust the level and tone of its output in response. (Lewitt describe it as "autofocus for your voice", although that »

oeksound

meet bloom, oeksound's adaptive tone shaper

Bloom analyzes the character of a signal and applies corrections to the perceived tonal balance for a more even and refined sound. Achieve warmth, clarity, or brightness with easy control for nuanced enhancements or striking transformations. Bloom unveils the full potential of your sound.

Learn more at oeksound.com

» could be taken to imply that it's varying the polar pattern, which is not the case.)

The way it works in practice is simple. Place yourself in front of the mic and you'll see one of the white bars on the zebra crossing illuminate, along with the Aura logo. Move around and the display will change to reflect your position, illuminating a lower, wider bar when you're near the mic and a narrower, higher one when you're further away. And if you listen to or record the output from the RAY as you talk or sing, you'll notice that the subjective level of your voice remains remarkably consistent as you move in and out.

The Low Down

To get a handle on exactly what Aura does, I tried playing test tones through fixed speakers and using my hand to trigger the proximity sensor. This showed that it varies the gain by perhaps 18dB or so between the nearest and most distant settings. Aura also applies what seems to be a variable low-shelving EQ to compensate for proximity effect. The most distant Aura setting kicks in when you back off to around a metre from the mic, and at this distance, the low end is about 12dB up compared with the closest setting, with the +3dB point at about 200Hz. Aura doesn't seem to change the tone in any other respect, which seems sensible, since there's no way of knowing how the inevitable increase in room pick-up will affect the character of the voice as the subject moves further away.

The zebra crossing display suggests that Aura works in fairly coarse steps, but that's a simplification to make the visual feedback more immediate. Sonically, it tracks distance very smoothly, with no discernible lag, and you won't hear any abrupt changes in tone or volume. It would take a lot of automation or a very intelligent compressor to recreate what Aura's level compensation does after the fact, and I don't think that even a multiband compressor can quite achieve the same degree of consistency in dealing with variable proximity effect.

The sensors follow what's directly in front of the mic, which can be an object as small as a human hand, so if you get too excited and wave your arms in front of your face, you'll likely provoke Aura into thinking you're nearer than you are. It might also be possible for a poorly placed music stand to cause issues, and you'll need to use the supplied pop filter rather than a conventional stocking-and-coat-hanger affair. But in general, it's foolproof, incredibly easy to use and remarkably effective. And, in case you were wondering, it doesn't seem to be affected by lighting conditions. I tested it in daylight and at night with all the lights switched off, and it worked equally well in both circumstances. My only quibble, really, is that if you move far enough sideways to be out of Aura's 'line of sight', it'll jump to the maximum distance setting, which perhaps isn't what you want. An option to retain the last actively detected distance in those circumstances might be preferable, not that you usually want to be moving that far off-axis during recording in any case.

RAY Of Sunlight

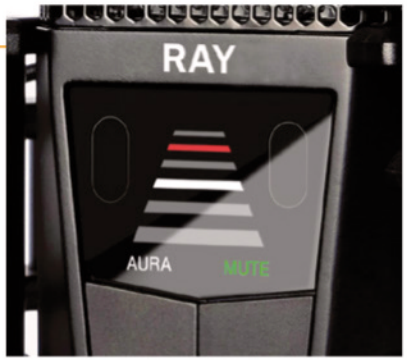
Aura's most obvious value is in live contexts such as streaming, broadcasting, podcasting and so on, but it definitely has the potential to be useful in music recording too. When you're moving a mic around to find the right placement, it's hard not to be swayed towards closer positions just because they're louder and more impressive for the same preamp gain setting. The RAY effectively levels that playing field, as well as offering a natural and transparent way of handling those performers who just won't stay still when they're doing their thing in the studio.

And in purely sonic terms, the RAY holds its own against other mics in this price bracket, with impressive specs that include an eye-catching 8dBA self-noise figure. Like some of the other Lewitt models I've heard, it has a relatively crisp and forward core sound, with a readily apparent presence boost, but is not so pre-equalised as to lose too much in the way of versatility. It sounded very good on my own voice, and the gain variations prompted by Aura did not introduce any detectable noise or other nasties. Lewitt have created a simple, effective and mostly foolproof solution to a genuine real-world problem, and it works beautifully. What's more, even with the Aura circuitry and sensors, it uses only 7.2 of the 10mA permitted in the IEC phantom power standard. I wonder what further creative uses Lewitt will find for the other 2.8mA...

The RAY package includes a discreet, magnetically attached pop shield that doesn't interfere with Aura's 'line of sight'.

Mute By Distance

Press and release the RAY's Mute button and it does what you expect, illuminating clearly in red to indicate that output from the mic has been attenuated by 70dB. Press and hold it, however, and you enter a mode Lewitt call Mute By Distance. The idea is that after you release the button, you place yourself at the furthest point from the mic where you want sound to be picked up. The appropriate bar on the 'zebra crossing' display will turn red to indicate that the RAY will auto-mute as soon as you get further away from the mic than that. This is another neat application for the proximity sensing, albeit one that's probably more relevant to live streamers and YouTubers than it is to music recording.

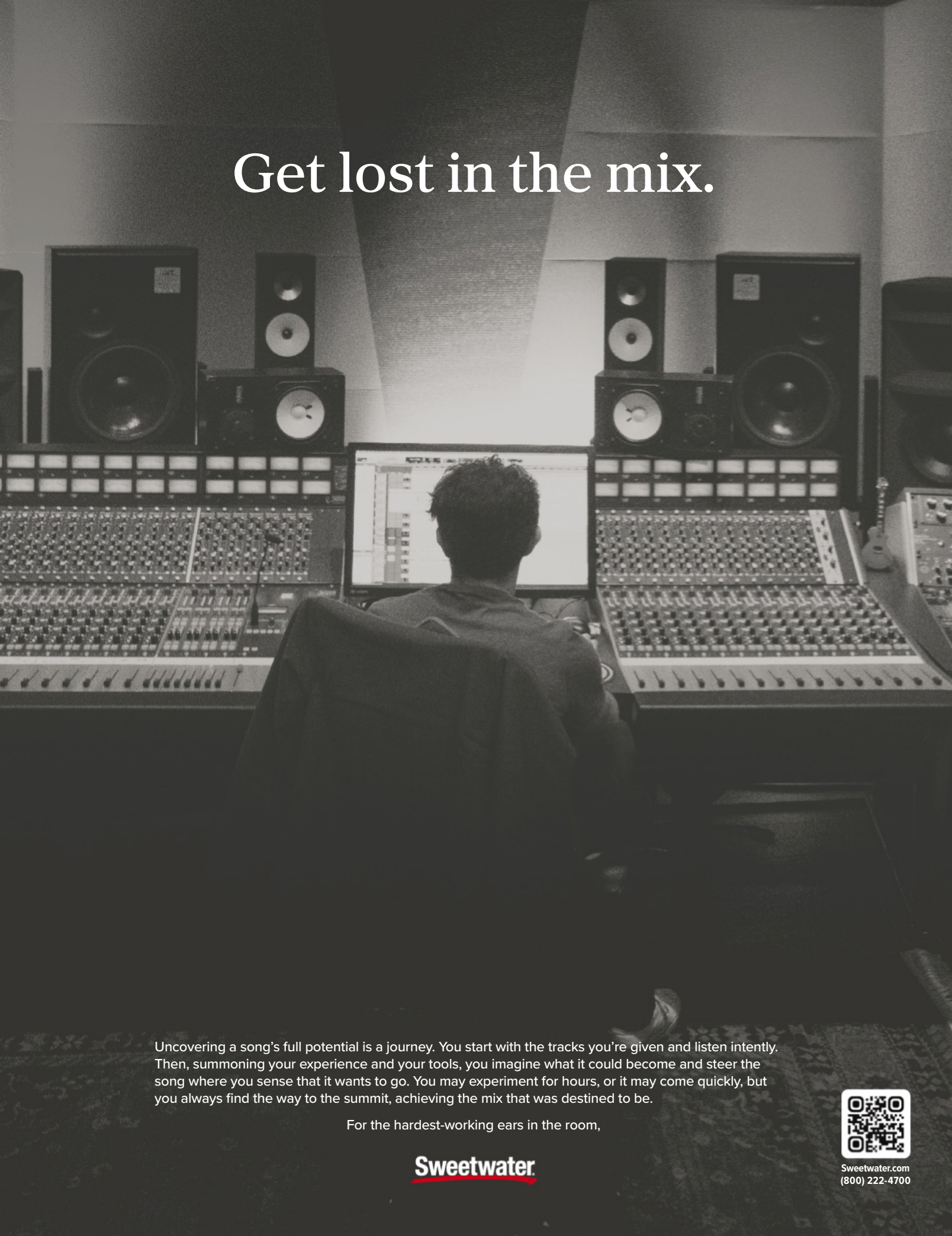


When Mute By Distance is active, the user sets a distance limit indicated by the red bar. Move further away than this, and the RAY will mute itself.

summary

Like all really good ideas, the principle of a distance-sensing mic sounds simple, but it's never been done before. Lewitt have got it right first time!

$ $349
W www.lewitt-audio.com


Get lost in the mix.
Uncovering a song's full potential is a journey. You start with the tracks you're given and listen intently. Then, summoning your experience and your tools, you imagine what it could become and steer the song where you sense that it wants to go. You may experiment for hours, or it may come quickly, but you always find the way to the summit, achieving the mix that was destined to be.
For the hardest-working ears in the room,
Sweetwater
Sweetwater.com
(800) 222-4700

Imaginando BAM Beat Maker & Music Maker

Music Production Software

This cross-platform package puts the fun back into music making.

JOHN WALDEN

Having access to vast collections of software instruments and effects, all housed within a top-tier music production DAW, is a truly great thing. However, it's also something of a double-edged sword, particularly if your latest musical inspiration fizzles to nothing as you try to decide between an endless set of synth presets, kick drum sounds or reverb choices. In terms of keeping the creativity flowing, sometimes having less choice means that you actually achieve more.

Which is kind of where software like Imaginando's BAM Beat Maker & Music Maker comes in. As a standalone, pattern-based music production system, it provides a compact feature set with a streamlined workflow. Fewer distractions, fewer decisions and — perhaps — more actual music. Well, that's the theory at least... so just what does BAM have to offer?

BAM Basics

The underlying concept within BAM's workflow is a familiar one. Essentially, you get a step-sequencer-based environment (up to 256 steps within a clip) featuring up to 16 sound sources. Projects are arranged in a scene-based song arrangement system, where each scene is able to contain an individual MIDI clip (pattern) for any/all of the 16 sound sources. You can then trigger these clips is various combinations or all the clips within a specific scene.

MIDI patterns/clips can be created in various ways. A Timeline view provides a simple grid editor that's great for drum pattern creation. There is also a Composer view that provides a piano-roll-style editing environment and is more suitable for melodic/chord instruments such as bass or keys. These views dominate the central portion of the BAM UI where, as well as the Timeline and Composer pages, the five buttons located far left also allow you to toggle between the Matrix (containing the matrix of MIDI clips organised into scenes), Automations (you can also step sequence automation of parameters at the clip level) and Mixer views.

While there are ways to integrate external sound sources into the BAM workflow, the software has its own suite of sound engines. These include some straightforward synth engines and a sample-playback engine, and part of the streamlined design intention is to keep BAM as self-contained as possible. The feature set includes a selection of effects that can be applied at the individual sound engine level plus two global effects that are accessed as sends from the individual sounds. Each sound engine has its own channel in BAM's compact mixer, which includes send controls, pan, volume and solo/mute buttons. The supplied factory content includes some useful samples, instrument and effects presets, and demo projects to help get you started.

It's also worth noting that BAM is cross-platform. I tried both the iOS and macOS versions and, while there are some technical differences (for example, the iOS version allows you to use external AUv3 apps as sound sources), the workflow is pretty much identical, and if you do confine yourself to BAM's internal sound engines, projects can be easily ported between platforms. MIDI export is also supported alongside audio rendering, and both can operate at the song or scene level of a project.

The Sounds Of BAM

BAM features 12 different sound engines. These include a number of dedicated '08' synth engines for kick, snare, clap, tom, conga, hi-hat and cowbell, each of which offers a compact set of controls to tweak the individual sounds. There is also a Drum Synthesizer engine with a more comprehensive control set that includes oscillator, FM and noise components to create a wide range of drum sounds.

The Oscillator and Hoffman engines provide synth-based options for non-drum sounds. Oscillator is a compact subtractive synth with dual oscillators, noise generators, ring modulation, FM, a multi-mode filter and saturation. Hoffman is a monophonic synth engine and is intended to do TB-303-style duties.

The other main option is the Sampler. This allows you to create sounds from a single sample, manipulate the sample in various ways, and automatically map it across the MIDI note range. You can use the factory-supplied samples or import your own. Samples used within a project are added to a 'pool' and the Sampler engine's Sample control lets you switch between these. This control can be automated, letting »

Imaginando BAM Beat Maker & Music Maker

€149

PROS

- Compact pattern-based beat and music creation.
- Easy to learn.
- Works on desktop and mobile.

CONS

- Primarily aimed at electronic music styles so not for everyone.

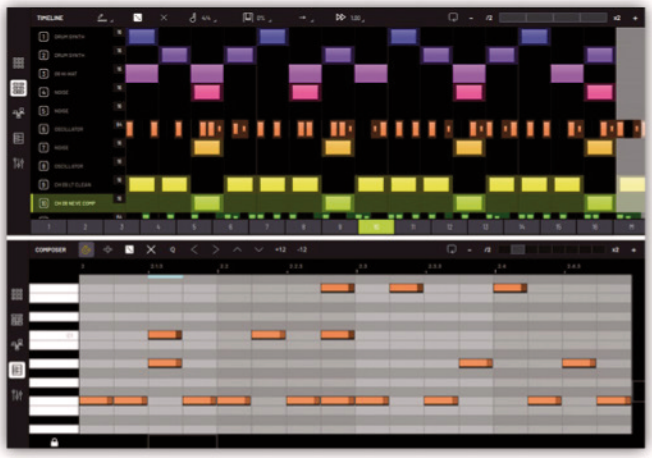
SUMMARY

Imaginando's BAM is a streamlined, pattern-based music production environment aimed primarily at electronic music production. The compact feature set strikes a good balance between depth and ease of use.

osmose

Next-gen standalone expressive synthesizer.
Add emotion & movement to your music with a simple touch.

expressivee.com/osmose



Toggling between BAM's five main windows, including the Timeline (top) and Composer (bottom) shown here, means you have a range of options for pattern editing.

» you use different sample-based sounds in different places within your overall arrangement if required.

Controls for the currently selected track/sound source are shown in the rack area in the upper half of the UI. The panels here include a compact version of the sound engine's controls and (far right) mixer/send controls. You can also add up to two effects devices per instrument from a selection that includes a filter, three-band EQ, parametric EQ, low/high shelf, chorus, bit reduction, stereo enhancer, delay, reverb, compressor and distortion. A Modulation panel allows you to configure multiple target parameters — for example, from the sound engine or effects — for either LFO or envelope modulation. Far left of the rack is the Trigger pane. This lets you set the default MIDI note (for example, for a drum sound) and, interestingly, offers a Probability control that can influence the likelihood of any MIDI note events within a pattern/clip being triggered or not, adding some very cool random variation to your patterns.

Project BAM

The 'beat maker' element of the title comes to the fore in the design of BAM's workflow. This is very much aimed at those who are happiest when building a track from patterns within a step sequencer. The Timeline panel lets you easily build a drum groove from multiple sound devices (kick, snare, hi-hat, etc) with a simple step sequencer view with your sound devices arranged down the left side of the display. Note entries here will then appear as individual clips for each instrument within the Matrix view.

For more detailed editing of individual clips (for example, for melodic instruments where you also need to specify pitch), the Composer panel provides a familiar piano-roll editor for the currently selected clip. Manual note entry here is very straightforward but you can also play MIDI in from a keyboard if you prefer. Again, you have a fairly typical suite of note editing tools for the task at hand.

As well as the LFO and envelope modulation options mentioned earlier, you can add step-based automation data to the currently selected clip via the Automations panel. Multiple automation targets can be specified via a very simple 'Learn' process and parameters adjusted on a per-step basis.

Each of the sound devices — including the Oscillator, Hofmann synth and Sampler shown here — provides a functional but suitably effective control set.

As well as giving you a clear oversight of your project, the Matrix view is where you can further arrange clips into multiple scenes. You get all the usual copy/paste/delete tools to do this, so building a song arrangement is conceptually very simple. Scenes (all the clips present within a horizontal row) can be triggered together via the Scene buttons positioned far right. However, you can also click on individual clips within this view to trigger them and they will simply replace any current clip playing for that same sound device in sync to the playback.

Small But Beautifully Formed?

Whether it's the sound devices, effects devices or some other element within the rack, the control sets are streamlined, but both functional and easy to navigate. However, tucked away inside each element are enough tools to get creative and keep things interesting, whether it's reversing the playback direction of a specific clip, generating random note data, or options to configure how the project moves through its various scenes.

How does the overall concept stand up? Well, it's certainly compact and anyone who has used a step sequencer before will soon find their way around the basics. Add a little time to appreciate some of the finer details and BAM can be quickly mastered. And, at that point, it will let you make your music without getting in your way.

The beat maker workflow has, of course, been around for some time so there are alternatives to consider alongside BAM. For example, you could adopt a similar approach within a mainstream audio+MIDI sequencer (if you can avoid getting distracted by their broader feature set), but software such as Korg's Gadget, with its similar clip/scene-based workflow and cross-platform (mobile and desktop) support, is perhaps a more obvious comparison. Gadget perhaps comes with a slicker look and more features but also, on the desktop at least, a higher price tag.

The step-sequencer workflow is undoubtedly a niche approach aimed primarily at electronic music styles but, in that context, Imaginando have struck a very interesting balance in BAM's design. The feature set has enough options to keep things interesting but is compact and constrained enough to easily master; BAM may not be the prettiest UI I've experienced, but it might provide just the constraint you need to focus on making music rather than constantly learning how your software works. Yes, there are some well-established alternatives but, if a compact, cross-platform solution appeals, then BAM is well worth a look. ■■■

$ €149, rent-to buy €14.90 per month.
W www.imaginando.pt

World class mics to capture your best.

Classic UA craftsmanship, designed to inspire singers, podcasters, engineers, and creatives of all types.

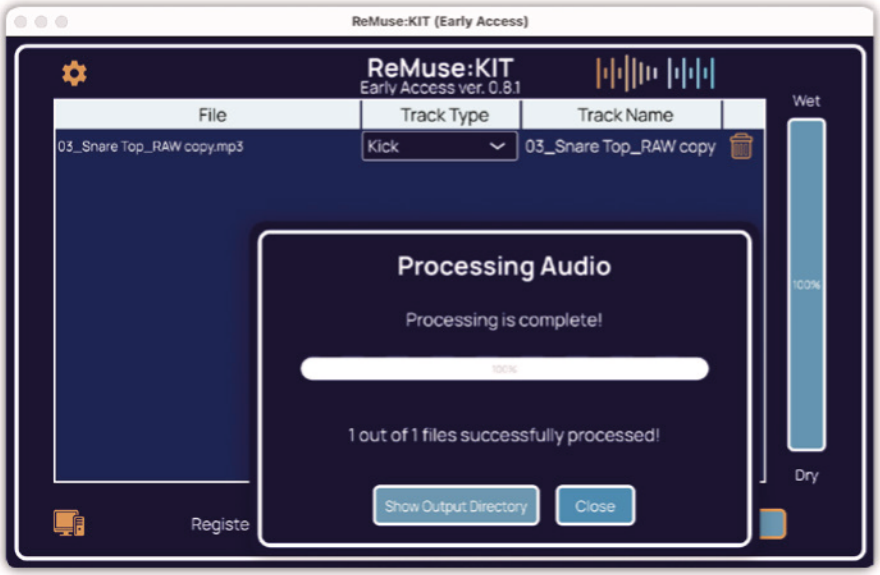
Learn more about UA microphones
uaudio.com/microphones

UNIVERSAL AUDIO

ReMuse ReMuse:KIT

Drum Separation Software

MATT HOUGHTON



Too much bleed on your close mics? Channel went down during your tracking session? This clever drum unmixing app could save the day...

ReMuse:KIT is a handy standalone application that appears to use machine learning to identify and then separate out individual drums from bleed, or even from a full drum kit recording — the idea being that you can extract and then either manipulate the individual kit pieces, or use them as triggers alongside an original stereo recording. It is compatible with Mac and Windows machines, and is available both to rent and to buy. You'll need an online connection for authorisation, which uses a serial number, but having done that once you'll always be able to work offline.

Overview

Operation really couldn't be simpler. First, you must import the drum file, and you can do this using a button and file browser, or simply by dragging the file from your OS's file browser onto ReMuse:KIT's GUI.

Next, using a drop-down menu, you tell the app what is in the recording — for example, it might be an 'image mic' of the whole kit (a drum mix, a room mic or overheads, for example), or it might be a close mic used on the kick, snare or toms and the file includes some bleed that you wish to get eliminate attenuate. Neatly, if a drum is named in the file, it will default to what's probably the right answer, which is a nice touch.

Finally, you decide if you'd like the app to perform a full or partial extraction: there's a wet/dry slider on the right for this. Then just hit Go. ReMuse:KIT will do its thing, which it does admirably quickly, leaving you with a new file. By default, this will be placed in the same folder as the first file imported into the session (I noticed that if I then imported a second file from a different location, the result would appear in he same folder as the first). Happily, there's also a settings cog on the app, and this allows you to define the folder where your results will be written.

In The Real World

So, operation is simple — but just how good are the results? I tested ReMuse:KIT on my M1 MacBook Pro, trying it out with several files of different types. And I started with the hardest: a full stereo drum loop, which I defined as an 'image mic'. Once ReMuse:KIT had done its thing, I imported the results into a Reaper DAW project alongside the original file and my first impressions were decent. It's important, though, to say that the extraction for this particular source wasn't *perfect*: there was a little 'echo' after the kick, for example, and the cymbals in particular sounded somewhat gated. Also, when playing all the extracted parts together, while they didn't sound terrible they certainly didn't null with the original. It might have been nice, I thought, to have an 'everything else' file... and on checking this point with ReMuse, I was told that this is already planned and will be coming in an update that may well be available by the time that you read this review.

Importantly, the results were very useful in practice. For instance, I was able to play the original track, and then have separate kick and snare tracks beneath it in Reaper, giving me the ability to gate, EQ and saturate the new standalone kick part (effectively parallel processing the kick) to create a very different composite sound. Similarly, I was able to use the snare track, with a little gating, to trigger a very believable reverb only on the snare, again changing the overall vibe of the loop. Alongside the original drum loop, this all sounded gratifyingly 'natural'.

Next up, I tried to use ReMuse:KIT to tackle some significant unwanted bleed on individual kit mics, where the bleed was causing problematic phase »

ReMuse ReMuse:KIT
$315

PROS
- Very effective on drum close mics.
- Potentially session saving!
- More features planned.
- Rental and purchase options.

CONS
- Can still leave you with some work to do.

SUMMARY
An effective way to extract virtual kit-piece mics from a stereo drum file, and an even more effective way to control spill on individual kit-piece mics.

Spirit
Stereo Pair

Origin
Stereo Pair

FREE!
2x SwiftShields

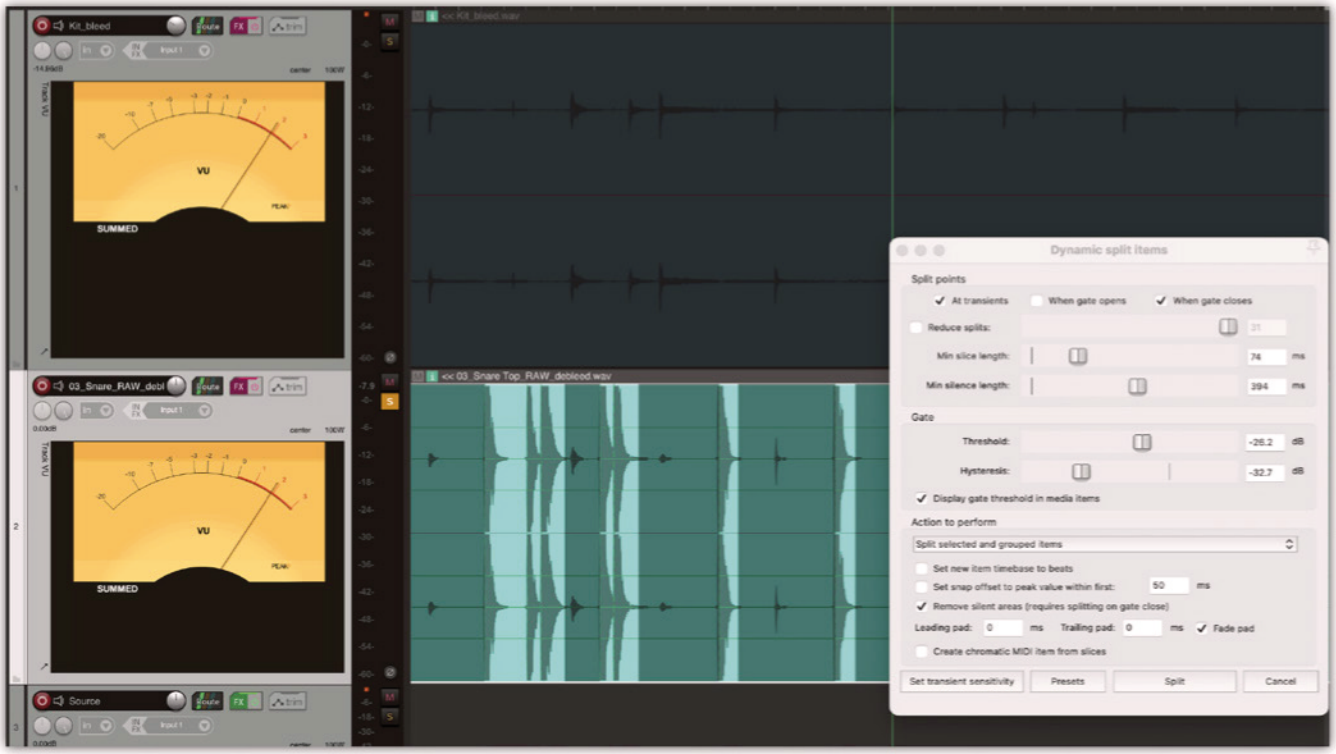
The power of two

Aston's legendary LDC mics, Origin and Spirit, are now available as stereo pairs. Both are bundled with a special-design stereo bar plus two FREE SwiftShield pop filter/shock mount sets. So not only does this open up a new world of lush, stereo soundscapes, but a host of other multi-mic applications too.

Double-up on the smooth, warmth of the Origins or the natural, sparkling detail of the multi-pattern Spirits, and you're all set to capture just about any sound source. We're talking world-class audio here, only without the earth-shattering price tag.

More at **astonmics.com**

ASTON MICROPHONES®



ReMuse:KIT works best on close mics, but it can extract individual drums from full kit recordings and loops too. The extraction in this scenario may not be perfect — but it's good enough that you can finish the job using more traditional editing tools, such as Reaper's Dynamic Split (pictured).

» cancellation with other mics used on other kit pieces, and this is where ReMuse:KIT really comes into its own. Dragging a raw snare mic track onto the GUI, ReMuse:KIT recognised that this was a snare part, and pre-selected that option. Hitting Go (with the processing 100% wet) I was soon presented with two files: one the dry, de-bled snare, and the other containing the bleed. Bringing these into Reaper, I could see and hear instantly that the results would be far more useful.

Focusing on the de-bled snare, the separation was impressive, though there were still ghost elements from the kick. Thankfully, these were now low enough that I could use Reaper's Dynamic Split (equivalent to Pro Tools' Strip Silence) to eliminate these unwanted elements — this didn't cause any unwanted change in tonality of the kick in either the kick mic or the overheads. I was left with a beautifully dry snare track that I could process as I wished to reinforce the overheads and room mic, and a separate track whose fader I could ride to remove or set the desired level of bleed. Often, I find that stripping all the bleed out of a snare track can cause havoc with the sound of the overheads, so it's great to have individual control like this. What's more, I then tried processing the same snare file but instructing ReMuse:KIT that I wanted to extract only the kick. It did this very successfully. So if, say, a kick drum mic went down on a recording session or gig, the chances are that you'd be able to extract a very usable kick trigger from another mic. I had similar joy separating kick drums and toms from bleed on their own mics, and that allowed me to get very surgical with a kick sound. I could boost some of the beater's attack, where doing that previously would have brought up all sorts of cymbal and snare detritus.

"I was left with a beautifully dry snare track that I could process as I wished to reinforce the overheads and room mic."

I should point out that if you do want to attempt the sort of gating or Strip Silence-style processing I've described above, you could come a cropper with more dynamic parts containing quiet hits. For example, if there are low-level ghost notes on the snare that you want to preserve, it will be harder to filter out any low-level remnants of the kick spill.

Extractor Fan?

ReMuse:KIT has so much potential, and I'm told further developments are on the way, including a new phase-alignment feature. Already, though, it could save your bacon if you have too much spill on a kick or snare mic and need more control over the sound, whether that be through processing or triggering samples to replace/reinforce the sound. Or if, as I said above, there's a problem on one of the main kit-piece mics and there's no chance to re-record. I wouldn't advocate that you allow your miking to get sloppy and get into the habit of relying on ReMuse:KIT, because it isn't perfect. But it *is* impressive, and in those situations where you have to work with what you've got, it could be a very handy problem solver.

$ Perpetual license £249.99 (about $315), discounted to £199 ($250) when going to press. Rental £9.83 ($12.41) per month for an annual subscription, or £10.99 ($13.88) per month for a monthly subscription.
W https://remuse.online

_AstroLab

ARTURIA
_The sound explorers

Go Beyond

Explore a vast spectrum of classic and modern sounds. AstroLab is a 61-key stage keyboard combining the power of synthesis, intuitive controls, and an innovative ecosystem - inviting you to focus on your creative expression and transition seamlessly from studio to stage.

Oeksound Bloom

Adaptive Tonal Shaping Plug-in

Oeksound's long-awaited third plug-in is a quietly radical alternative to equalisation.

SAM INGLIS

Some plug-in developers seem to release a new product every week. Others are more selective — and they don't come much more selective than Oeksound. Launched in 2018, the enigmatic Finnish coding wizards' first plug-in quickly became one of those tools that every big-name engineer seems to have in their kit, and for good reason. In an era when we often work with phone demos and ropey home-recorded tracks alongside pristine studio material, Soothe's unique ability to dial back unpleasant resonances and alleviate harshness makes it invaluable.

The success of Soothe perhaps meant that Oeksound's follow-up, Spiff, slipped under some people's radars, but as I've got more of a handle on how it works, it too has become one of my favourite dynamics processors. It offers a novel, frequency-dependent approach to transient control which is completely different from other 'transient shapers'. Since then, there's been a version 2 of Soothe, and a low-latency derivative optimised for live sound, but it's been more than five years since Oeksound launched an entirely new plug-in.

Oeksound Bloom
$209

PROS

- A novel process that is universally applicable in mixing, mastering, restoration, broadcast, you name it.
- Has a subjectively positive effect on almost any sound, even with minimal control input.
- Capable of transforming the timbre of a source in ways that simply aren't possible with other tools.

CONS

- It can be hard to know when you've gone too far.

SUMMARY

Bloom is a truly remarkable plug-in that can 'improve' the sound of recorded audio in ways that were not previously possible.

At this year's NAMM Show, however, Oeksound finally unveiled their third major product. And, spoiler alert, it was worth the wait.

What It's Not

Oeksound describe Bloom as doing "What we wish an EQ would do," and this, it turns out, isn't just one thing. Like an EQ, it has corrective uses, but also creative applications, allowing the timbre of recorded audio to be modified. However, Bloom is not an EQ. Nor, despite similarities in the user interface, is it a multiband compressor. And although it has something in common functionally with some products based on machine learning, such as Sonible's SmartEQ, it's a purely algorithmic design. So what is an "adaptive tone shaper" when it's at home?

There's a certain level of mystery surrounding exactly what Bloom does, but a few things are clear. Firstly, unlike Soothe, it's essentially a broad-brush process: rather than notching out hundreds of tiny resonances, the tonal changes it applies typically span an octave or more. Second, it's a dynamic process, in the sense that it is constantly adjusting what it does in response to the source. Third, it can also be dynamic in the sense of changing signal dynamics, but unlike a compressor, its effect is for the most part level-independent.

Bloom is a native plug-in available for macOS and Windows in all the usual formats. It's authorised using the iLok system, but a physical dongle is not required. It occupies a relatively compact window tinted an attractive shade of rose pink, with no extra tabs or panes, and there are just five main controls. The large Amount control is self-explanatory, but the four Tone Control sliders that adjust specific frequency ranges are less so. For one thing, they're actually X/Y pads rather than conventional faders: moving the central handle up introduces a 'boost' in that signal range, and pulling it down initiates a 'cut', but you can also drag the handles sideways to set the centre frequency of the band. This makes it seem superficially like a multiband compressor, but in practice, it's very different. »

MONOLITH

BY RICHARD HARVEY

ORCHESTRAL TOOLS

I've placed the terms 'boost' and 'cut' in inverted commas because it's important to understand that they don't apply gain changes in any normal sense. It would be more apt to think of them as offsets that can be applied to the core process: the settings of the sliders adjust the 'target' response that Bloom tries to nudge your sound towards. Crucially, leaving all the sliders at zero doesn't mean that no processing takes place: it means that the processing will aim to push your sound towards a balanced target. By contrast, pushing the high and low sliders up and the middle ones down will define a 'scooped' target, and so on.

Line Dancing

Bloom's processing is represented in an animated 'processing graph' which occupies the space beneath the Tone Control sliders and has amplitude on the vertical axis and frequency on the horizontal. Turn the Amount dial right down to zero, and the graph looks like a flat line. As you turn the Amount up, the line begins to deform and move around in response to audio input. Where Bloom's algorithm decrees that a boost should be applied in a certain area, this shows up shaded white, while cuts are indicated by the line falling below the horizontal and eating into the solid mauve region. Additional cuts or boosts prompted by the slider settings are shown in the colour of the relevant slider. Draggable handles at either end of the processing graph allow you to introduce low- and high-pass filters.

The Amount control ranges from zero to 10. As you push it up, Bloom becomes increasingly assertive in the boosts and cuts that it applies, but the overall subjective level remains constant. A Wet Trim setting lets you adjust the level if for some reason this doesn't happen. This is also useful if you want to employ Bloom as a parallel processor, and when you push the Amount control beyond 7. At this point, the white boosted areas on the processing graph start to hit the ceiling, introducing a cool and easily audible compression effect. Again, this isn't the same as conventional broadband or multiband compression, but it imparts a similar sort of pumping, breathing quality to the signal. This part of the processing is level-dependent, and a Squash Cal control has a function somewhat similar to that of the threshold setting on a normal compressor.

The Attack and Release controls govern the speed with which Bloom reacts to changes in the audio. Like most Bloom parameters, they are arbitrarily calibrated from zero to 10, and it would probably be an oversimplification to call them 'time constants'. Their settings are often more obvious from the movement of the processing graph than from the actual sound; at least until you reach the squash range, it remains smooth at all settings. Bloom can operate in standard or high-quality modes, the latter incurring a greater CPU overhead but making an audible difference on some sources, especially complex material. It can also be switched into a low-latency mode for tracking, which reduces the look-ahead delay to 1.33ms at 48kHz at the standard quality setting.

Finally, if you instantiate Bloom on a stereo track, it defaults to applying the same processing to the left and right channels, presumably responding to the sum or average across both. However, it's possible to switch it into Mid-Sides or dual-mono modes, and if you do this, each Tone Control band sprouts a second slider, allowing you to set different targets for left and right, or for the sum and difference channels.

On stereo tracks, Bloom gives you the option to process left and right or Middle and Sides signals independently.

Rose Tinted

If describing what Bloom does isn't easy, then neither is describing how it sounds. In the most general terms, the best I can come up with is to say that it acts like a magic mirror, reflecting back a version of the source sound that is in some indefinable way more attractive. Or perhaps it's the audio equivalent of the filters that make us look younger and more beautiful in social media videos. If you're coming at it with a mindset formed by EQ and multiband compression, it takes a little while to get used to the fact that Bloom can have a powerful effect even with all the Tone Control sliders at zero. Simply insert it on a track and you will hear an immediate change in the sound — one which is likely to make you think "Wow". Within two minutes you'll have forgotten it's there, and it's only when you bypass it once more that the harsh reality of your recording comes crashing to your attention.

In theory, if you feed Bloom an unchanging signal that's already well balanced by its own lights, any movement on the processing graph should be minimal, or at least equally spread across the spectrum. To get a handle on what it does, I tried using pink noise as a source, since some people recommend this as a balanced target spectrum for mixing. Doing so made clear that Bloom's idea of a balanced tonality is more 'scooped' than the spectral balance of pink noise, because it tried to push up the bass and

MASTERING... SOLVED.

Michael Romanowski, Coast Mastering
(Alicia Keys, Too $hort, America...)

- **BDA4M 4 CH DAC with switchable BX5s**
- **BAD4M 4 CH ADC with switchable BX1s**
- **B16-BMB3 Dante**

coast mastering

"It gives me flexibility...and clarity that I need in my work."

"We did a shootout between 6 different manufacturers and the BURL was the clear leader...Great bass and perfect top end."

Bill Skibbe, Third Man Mastering
(Jack White, The Black Keys, The Kills...)

- **BDA8 8 CH DAC with BOPA14s**
- **2 BAD8 16 CH ADC with BX1s**
- **B80-BMB1 Digilink**

THIRD MAN MASTERING

GET IT RIGHT THE FIRST TIME.

BURL

BURLAUDIO.COM/MASTERING

MADE IN THE USA

» high frequencies and make a broad cut in the midrange. At the same time, thankfully, its target is clearly much less bright than white noise.

However the target is defined, it works, because I don't think I have ever encountered another plug-in where the default preset was so universally applicable. As soon as you instantiate Bloom, you'll hear your audio being subtly massaged towards Oeksound's target frequency balance, and it's rarely the worse for it. Whatever this target balance is, it works equally well on anything from individual sources to the master bus, and from delicate acoustic recordings to brutal electronic kick drums.

As I've already mentioned, the process is incredibly smooth, and unless you stray into the 'squash' range, you're unlikely to notice any changes to the dynamic behaviour of your signal. On a drum loop, for example, the audible effect of the Attack and Release controls has more to do with changing tone colour than with bringing up or down the sustain portion relative to the transients.

Flower Festival

The applications for this plug-in are endless, but I'll list a few highlights that I encountered in my testing. First of all, it's spectacularly good at dealing with damaged or badly recorded audio. In restoration work, it very often achieves in seconds what you could spend hours trying and failing to do with EQ or multiband compression. Bloom is at its most impressive in those apparently impossible situations where you need to reshape the tone of a track or source but doing so with EQ seems to bring out all sorts of nasties that the wonky timbre previously obscured.

Second, it's a remarkably effective alternative to equalisation for tonally variable sources like the human voice, and especially voices afflicted by bad room sound, poor mic technique or inappropriate mic choice. The default target profile generally drags raw voice recordings towards a brighter sound, filling out the upper midrange whilst controlling wooliness and proximity boost in the low mids. Yet, when Bloom encounters sibilants and other consonant sounds that already have plenty of high-frequency energy, its adaptive nature means these are left well alone or even reduced. The upshot is that you get a more tonally consistent vocal sound, with no need for de-essing and no risk of introducing side-effects such as lisping.

Third, it often enables you to apply a greater degree of tone shaping than is possible with other tools, whilst remaining natural. For example, I was able to take a drum overhead track recorded with a vintage ribbon mic, push up the High and High Mid Tone Control sliders and produce something that sounded for all the world like it had been tracked with a capacitor mic. Attempting a similar transformation with EQ just made everything sound, well, equalised. Or rather, badly equalised.

"Bloom is probably the closest thing I've come across to the proverbial 'make it sound better' plug-in."

Fourth, Bloom naturally saves you from yourself in a way that an equaliser won't. I'm sure we've all had the experience whereby we boost something a little bit with EQ, like the results, crank the boost a little higher, and end up losing our mental reference as to what that source or mix should sound like. The consequence is often an instrument or mix that sounds harsh, brittle, thin, boomy or otherwise unbalanced, and doesn't translate well between systems. Because Bloom is always pushing things towards a reference spectrum that is intrinsically balanced, it's much harder to fall into this trap.

Wish Fulfilment

In short, then, Bloom is probably the closest thing I've come across to the proverbial 'make it sound better' plug-in. Are there no down sides? Well, there are certainly sources where static EQ is preferable to my ears: on heavily distorted guitars, for example, Bloom seemed to lose some of the solidity and substance of the sound. And there will always be times when you need to use EQ to deliberately make something sound pokey, aggressive or otherwise unbalanced; using the Tone Control sliders can force Bloom into adopting an unbalanced tonal target, but it won't deliver the bite you get from pushing the midrange on an API. The 'squash' effect is interesting and characterful, but I didn't find a real-world use for it during the review period. And there were times when I wanted to be able to process only part of the frequency spectrum, which isn't currently possible.

Most of all, though, because Bloom's processing is so seductive, it's alarmingly easy to apply more than you need to. Like Soothe, it almost always makes whatever you're applying it to sound nicer in and of itself; but that doesn't always mean it works better in the mix. A timbrally balanced mix isn't normally achieved by making every individual element timbrally balanced, but by playing the contrasting tonal imbalances of different sources off against each other, and there are times when sounds need to be unbalanced and harsh in order to fulfil their role in the bigger picture.

Hence, as I experimented with Bloom, I became aware of a slightly paradoxical aspect to its operation. On the one hand, it is uniquely transparent, in that it can make timbral changes sound natural that would be impossible with EQ or other tools. On the other, that means it can actually have a sound of its own, in that if you Bloomify too many of the sources in your mix, it will begin to acquire a sort of homogenised, 'too good to be true' quality. This outcome is rarely something you're consciously aware of, more an uncanny feeling that lurks in the back of the mind, and as such, is easy to overlook.

But at the end of the day, what this boils down to is really just that it's possible to over-use this plug-in — and if that's a criticism, it's one that applies to every plug-in ever made. Self-restraint is needed to apply any kind of audio processing, and if Bloom requires more self-restraint than most, it's because its effect is so appealing.

Equalisation is one of the most fundamental tools available to the audio engineer, and it's something we take for granted in tracking, mixing, live sound, restoration, mastering and every other context. Oeksound have set out to make Bloom the processor they wish EQ was — and, consequently, it has an equally broad range of possible applications. This is a unique plug-in that has something to offer everyone from broadcast engineers to film dubbing mixers to mastering engineers, from the most humble of home studios to the most advanced mixing suites. Bloom really is pretty special, and if you don't believe me, sign up for the 20-day free trial!

$ $209
W www.oeksound.com

Huge Sound.

Introducing the LiNTEC Vintage Program Equalizer.

The most renowned studio EQ ever known is now within your reach! With the LiNTEC, you'll sculpt your tracks hands-on via the classic Pultec-style workflow. Bring air and space to vocals and stringed instruments, beef up the low-end of kick drums and basses, or add a touch of warmth and weight to an entire bus. Once you've run your tracks through a LiNTEC, you'll wonder how you got by with software EQs for so long. And you get all of this for a lot less than the price of other classic hardware EQs.

Visit **www.lindellaudio.com** and get the full story.

Lindell Audio

A Rad Global Distribution company

©2024 All Rights Reserved. Lindell Audio.

–Win! SSL Fusion– & THE BUS+

Worth $5394

For this month's exclusive *SOS* competition, we've teamed up with legendary studio brand Solid State Logic to offer you the chance to win not one, but two of their acclaimed analogue processors.

First up is THE BUS+. SSL made their name in the studio world with the groundbreaking G Series consoles, and a key part of their sound was the VCA compressor built into the stereo mix bus. This bus compressor was renowned for adding 'glue' to a mix to bring all the parts together into a cohesive whole. It became so popular that SSL released a number of standalone hardware versions, and even a software plug-in emulation. THE BUS+, however, takes things a step further, by combining the classic circuitry with a number of extra features never before seen on an SSL bus compressor.

So, in addition to the comprehensive compression controls, you also get a wet/dry blend for parallel compression, plus a range of stereo processing modes including dual mono and Mid-Sides. There are also extensive side-chain filtering options, plus the ability to use an external key signal, and even a side-chain send, allowing you to patch another processor into the side-chain path. It also lets you switch from the classic SSL feed-forward sound to Feedback Mode, for gentler compression. A button marked 4K introduces some even-order harmonics to evoke the sound of SSL's classic mixers, and some clever use of the front-panel buttons lets you dial the dirt in, from super-clean to heavily coloured. Then there's the inclusion of a two-band dynamic EQ, which features adjustable filter types, time constants and frequencies, and which can be placed before or after the compressor in the signal chain, as needed.

To enter, please visit:
https://sosm.ag/ssl-comp-0524

And that's not all. The lucky winner will also receive an SSL Fusion: a powerful, stereo analogue coloration processor for the mix bus that has no fewer than five processing stages. Vintage Drive is a thickening saturation effect, Violet is a two-band EQ, HF Compressor tames the high end, with adjustable crossover, Stereo Image is a Mid-Sides processor, and there's an output transformer.

To be in with a chance of winning this fantastic duo, simply follow the URL shown, and answer the questions there by Friday 7th June 2024. Good luck!

Prizes kindly donated by Solid State Logic
W www.solidstatelogic.com

"It's the only dynamics processor you'll need on your mix bus."

APOGEE

Apogee Control Remote*

Mix Atmos With Ease

Introducing the 16x16 Special Edition

Our Highest Performing System...**Ever.**

A new standard has been set.
Learn more at apogeedigital.com

*Remote sold separately

SWAM String Sections combines playability with microscopic levels of control over the sound.

Audio Modeling SWAM String Sections

Modelled Orchestral Strings Instrument

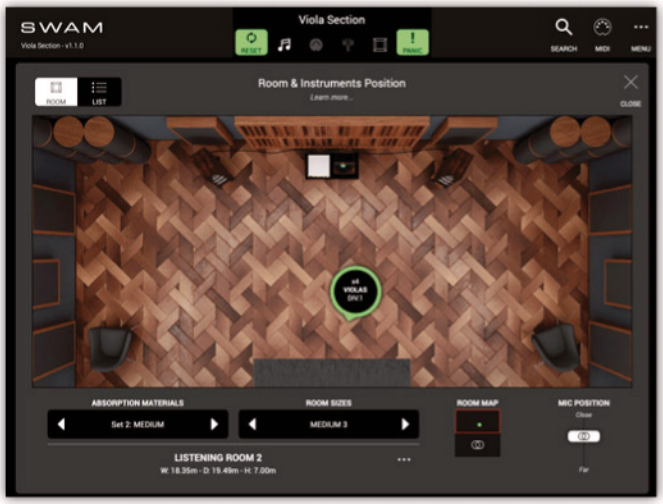
SONAL D'SILVA

If someone had told me when I was a teenager that there would come a day when you could 'play' a string instrument in real time, in your DAW, with just a MIDI controller, and that it had a small hard-disk footprint plus hassle-free installation, I'd have thought they were a bit mad. Back then, virtual instruments — string or otherwise — were characterised by somewhat accurate representations of the sound they were trying to emulate, dodgy GUIs, and the need to load a different patch if, heaven forbid, you wanted multiple articulations in the same piece. Sophisticated sample libraries improved the landscape greatly, and then came the physical modelling of instruments, designed for real-time performance and maximum control over as many parameters as possible to make the virtual instrument feel like the real thing. It's mind-boggling, quite frankly.

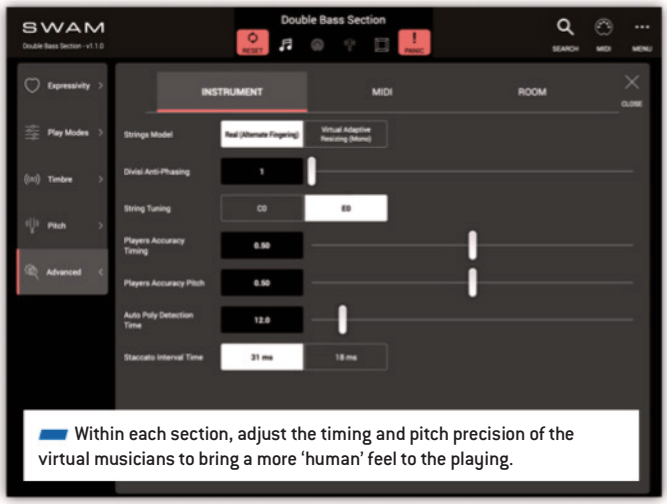
The Power Of Physical Modelling

A leader in this space is the Italian audio software developers Audio Modeling, who previously brought us Solo Strings, and recently released SWAM String Sections, powered entirely by their SWAM technology. If you're familiar with physical modelling vs sample-based libraries, you already know the drill, but anyone who's new to this, let's take a moment to explore the difference. Sample-based libraries offer pre-recorded samples of real instruments played by musicians; you can get beautiful, 'sounds-like-the-real-thing' samples to work with, but you've largely handed over control of phrasing and expressiveness to the musicians who played on the recording. A good library will fulfil your needs in terms of articulation and dynamics options, but you're still choosing from an existing palette and that may miss the target when it comes to the expressiveness and emotional resonance you're after — when it comes to music, there's more than pure technique that makes a piece what it is.

The possible limitations of a sample-based library can be best explained in sound design terms: say you have a large library of footstep sound



In the Room Simulator view, choose a virtual room, adjust mic proximity, and change the placement of your instrument sections by clicking and dragging in real time.



Within each section, adjust the timing and pitch precision of the virtual musicians to bring a more 'human' feel to the playing.

effects to choose from and you narrow it down by shoe type (heel), surface (marble) and speed (slow). The element most likely to not match your intent is performance. You want the slow, confident walk of a woman exploring her luxurious new apartment; the sound effect from the library hits all the marks (walking on a marble floor slowly in heels) but you can't hear the confidence — it just sounds like someone trying to be quiet and failing because, well, heels. This is the difference performance can make, be it in a sound effect, or an instrument sample. A virtual instrument built using physical modelling adds a highly-controllable performance element to the mix, allowing you, the composer, to 'perform' the instrument in real time, which adds a whole new dimension to the sound. (For a deep dive into the technology, check out this *SOS* article from way back in 1997: sosm.ag/4clJdAX).

This technology makes a difference to your experience in two other obvious ways: storage and performance. SWAM String Sections' installation is fuss-free and barely makes a dent in terms of hard disk space (approximately 430MB for the complete bundle with all plug-in formats), while RAM usage is about 350MB per instrument instance. The instruments support Audio Units, VST, VST3, AAX 64-bit, Native Instruments Komplete Kontrol, and run as standalone instruments as well.

The Instruments

SWAM String Sections contains a suite of four separate plug-ins, each dedicated to a group of instruments from the string section of an orchestra — Violins, Violas, Cellos and Double Basses. The Violin section offers an ensemble of four to six players; Cello, Viola and Double Bass offer three to five players per divisi. This allows the composer to build orchestras of different sizes, ranging from chamber to symphony (CPU Gods willing).

All four instrument sections require control of the Expression parameter, so the first step is plugging in your MIDI controller and making sure the parameter is assigned to a strip/fader that you can play comfortably. (Some composers assign dynamics to a breath controller.) This parameter is key to being able to play the instruments; in fact, you won't be able to generate any sound at all if you're not set up to control Expression. The next step is to set a healthy monitoring volume so you're not riding the Expression slider too hard. The user manual makes a special note of this very early on and lets you know that the slider will go red and warn you if the level of Expression stays above 75 percent for too long. After that, set up vibrato control and you're off to the races.

Each section is meant to emulate a real ensemble with each player playing a slightly different instrument, leading to the variations in timbre, intonation and timing that give a section that 'human' feel. However, phase issues are inevitable when using a virtual instrument, so when it comes to placing multiple instances of the same section in one project, pay attention to the Divisi Anti-Phasing parameter, and adjust to prevent phase-related artefacts.

Any discussion about the tone of the instruments on offer will be subjective because it depends on how you like things to sound, and also on how skilled you are at coaxing sound out of the instruments. Unlike using a sample-based library, you can't separate yourself from the sound coming out of the SWAM String Sections instruments. With great playability and control come a great many parameters that have to be skilfully manipulated and it is definitely something that requires practice. That being said, the violas do well in their lower registers; the cellos are evocative with the right expression and vibrato; the violins are to be handled with care in the higher registers (things can get shrill on longer notes); the pizzicato double basses are immediately fun.

Early users of SWAM String Sections will be pleased to note that in the updated version, the ambiguous Players Accuracy knob has been replaced by two settings that enable you to individually adjust timing and pitch precision in order to introduce more variability to each player's performance, »

Audio Modeling SWAM String Sections

$500

PROS

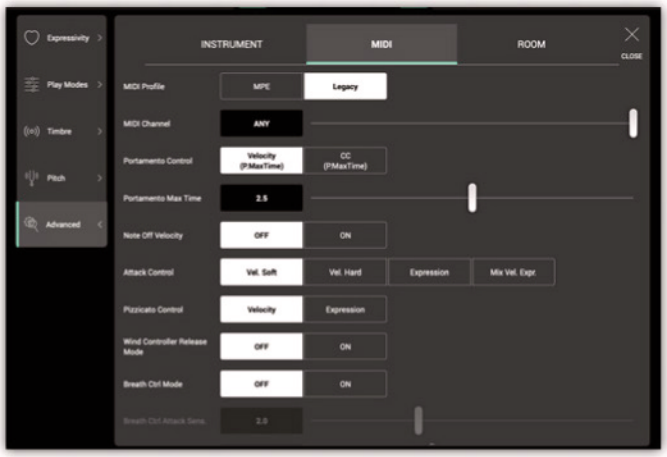
- Real-time performance of virtual instruments.
- Highly detailed options to enable control and expressivity.
- Excellent, easy-to-use GUI.
- Hassle-free download and installation.
- Minimal hard-disk footprint.

CONS

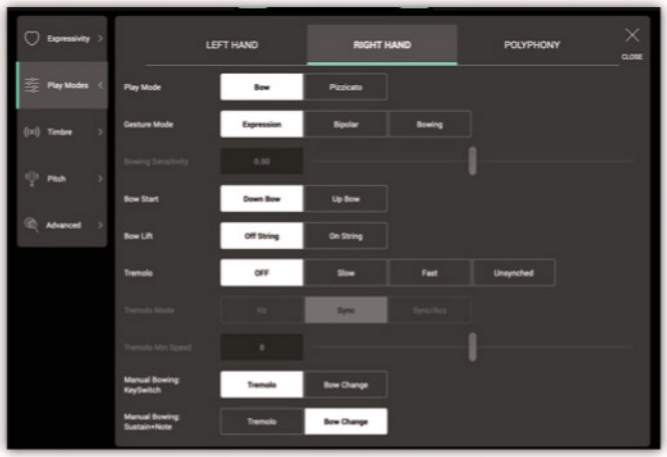
- Steep learning curve for newbies.
- High CPU load with multiple instances of the instruments.

SUMMARY

SWAM String Sections rewards time invested in learning it and the flexibility and control it offers to shape tone, dynamics, and performance means it could be a valuable addition to your arsenal of composition tools.



The threshold between portamento and legato is controlled using the Portamento Max Time parameter, with the option to disable portamento entirely by setting it to Off.



All the parameters you need to control tremolo rate in one place.

» which is responsible for that loose feel that you get when a group of musicians plays together. The settings can be found in the Advanced menu.

Articulation & Expressivity

An issue that kept cropping up was how tricky it was to avoid a portamento articulation when playing legato. This can be controlled either by adjusting the Portamento Max Time parameter (setting it to Off disables portamento entirely), or by remapping the velocity curve to adjust the sensitivity. For articulations like détaché, a sustain pedal is required; spiccato and flautando are possible to perform by adjusting bow lift and bow pressure parameters; some will be disappointed to hear that col legno is not available. Tremolos seem to sound more natural when the Unsynchronized option is selected, instead of Sync (which synchronises the tremolo rate to the current project bpm); the Sordino is so subtle, it barely makes a difference to any real expressivity.

Bow pressure, bow position and bow lift are all adjustable, further helping to shape dynamics and tone; also on offer are keyswitch-assignable harmonics controls, and an alternate fingering menu that lets you select the position of the left hand on the fingerboard. The manual has a clear and helpful guide on how to perform the various articulations and this deserves a shout out because a lot of software user manuals, while intending to be helpful, are most definitely not always clear.

Room Simulator

What good is a realistic-seeming virtual instrument if you have no sense of the space in which it is played? The folks at Audio Modeling have addressed this issue by creating the Room Simulator. Each instrument section comes equipped with the ability to choose the environment in which it is played, and rooms can be chosen based on absorption characteristics of materials and room size. A handy set of presets lets you place the instruments in rooms like Listening Studio (medium absorption, medium size), Cathedral, Concert Hall and Church... you get the idea. You can also adjust exactly where in the room the instruments are placed by changing distance and angle, and select microphone proximity.

A notable feature is that independent instrument sections 'talk to each other' about their room positions. For example, if you have cellos, violins and violas loaded on separate tracks, you can see the position of all the instrument sections in relation to each other by clicking on any one of the interfaces; modifications to all instrument positions are also possible from any one interface. The room selection is global, so a change to the type of room on one track changes the room type for all the other instrument sections — obvious, really, because you do indeed want all your instruments to be in one location.

A special shout out to the architects of SWAM String Sections for going the extra step and making the instruments user-friendly for blind and visually-impaired musicians via menus that are accessible to screen readers.

Conclusion

There is no doubt that the learning curve for shaping the sound of SWAM String Sections is steep. There are a lot of parameters that you are able to control, and being able to control them simultaneously is where practice is key. It takes a different kind of skill to make the instruments sound musical, especially if you're not an experienced performer or don't come from an orchestral background.

Your assessment of the sound on offer will also depend on the genre of music you're composing for, and if accuracy and realism are important to your work, you will perceive the SWAM String Sections differently than if you work in a genre or on a project that allows for a more experimental use of sounds.

Also, knowing what contributes to making the instrument sound 'real' and not like a synth is really the first step. If you're new to orchestration, it's a good idea to go down the rabbit hole of techniques and arrangement best practices, just to understand what you're aiming for. Doing this might save you the pain of hours spent trying to fix it with EQ because 'somehow it just doesn't sound quite right'.

Is the goal to replicate the sound of a real string section? Or is it to utilise this as a tool in your sound library? For every composer/musician that complains about the lack of realism of the sound, there is another composer/musician that can't believe they can tweak this many parameters and work with orchestral ensembles without leaving their bedroom studio. Simplistic? Maybe, but eventually it all starts with one composer sitting down (is anyone using a standing desk?) and reaching for the tools at hand to create a piece of music that a listener responds to. SWAM String Sections offers immense possibilities and, in the right hands, is a great tool to make your musical ideas come to life.

$ $500
W www.ilio.com/swam-string-sections
W www.audiomodeling.com

1073SPX-D

THE WORLD'S FIRST **GENUINE 1073®** INTERFACE

1073 Channel Strip - 9x16 Audio Interface
USB Connectivity - ADAT Expansion

"The brilliant 1073SPX-D houses the world's most iconic input module alongside high-definition AD/DA converters. It's my go to channel strip in the studio and the only hardware I need for a remote session."

John O'Mahony, Coldplay, System of a Down, Kaiser Chiefs

Learn More by scanning
the QR code or visit neve.com

Neve

DESIGNED AND CRAFTED IN THE UK BY NEVE® ENGINEERS

The top part of the GUI, which resembles the hardware, is all you see on first opening this plug-in, and you can do a lot with that. But pop open the control pane and you can do so much more — there are some really thoughtful features here.

Tone Projects Hendyamps Michelangelo

Analogue-modelling EQ Plug-in

Tone Projects' range may not include many plug-ins yet — but they're all up there with the very best.

MATT HOUGHTON

Back in 2004 Rune Lund-Hermansen gave us Otium FX's Basslane, one of the first low-frequency stereo width plug-ins, and I got a lot of use out of it! Two decades on, his company is now Tone Projects and they offer just a few plug-ins, but they're some of the very best analogue-modelling processors I've used. In our *SOS* October 2020 review (https://sosm.ag/tone-projects-unisum), Eric James described their Unisum as a "unique, stellar-sounding compressor". Then came their Kelvin 'tone shaper', which we've not reviewed but I can confirm is a wonderfully versatile, great-sounding dual-stage saturation processor with a neat pre-/post-emphasis EQ. They've also reworked Basslane, giving it lots more features.

But for their latest plug-in, they've taken on something very special — and they've done a cracking job!

Tone Projects' Hendyamps Michelangelo is officially endorsed by Hendyamps, whose hardware is a very high-quality stereo valve EQ and saturation processor. I had the pleasure of playing with one for a day or two and enjoyed it immensely. There aren't many controls, but using it isn't always straightforward because the bands interact and the saturation can be seductive (it's easy to overcook things). But used judiciously, it's a wonderful thing indeed. This plug-in mimics the hardware in obsessive detail, and it's one of the most convincingly analogue-sounding plug-ins I've used to date — but compared with the hardware, its functionality has been beefed up considerably. It's available for Mac and Windows hosts that support AAX, AU or VST3 plug-ins. Authorisation is by serial number, and installation quick and easy.

A Chip Off The Old Block

The default GUI resembles the hardware, but a pop-out 'advanced' panel offers many more options. More on that later, but there are useful extras on the basic GUI too. Typical facilities including undo/redo, presets, bypass and A/B comparison buttons are joined by an EQ scale control: you can exaggerate a curve, scale it back, invert it and even exaggerate that inversion; the range is ±200%. Input and output level controls allow you to get the incoming signal in the sweet spot and set the output accordingly. You can also set the processing quality from Low Latency (a decent approximation for tracking) to Pristine and, whatever the playback quality, you can set the plug-in to render at Pristine.

A real-time Autogain facility is convenient, but more accurate is the Match button. Hit this, play audio through the plug-in, and it calculates a more precise

Tone Projects
Hendyamps Michelangelo
$249

PROS
- Exceptional quality of analogue modelling.
- Many more features than the hardware, making it much more controllable.
- Dynamic EQ, transient/sustain EQ and M-S balance are genuinely useful options.
- The basics are easy to grasp quickly.

CONS
- If you're to get the most out of it, there's a learning curve.

SUMMARY
Hendyamps' Michelangelo is a wonderful beast and Tone Projects have tamed it — without breaking its spirit!

adjustment. It's a very helpful feature for mastering and stereo bus work. I don't normally use gain-matched EQ when working on individual tracks in a mix (if I want to nudge up the meat of a snare, I probably don't want the higher frequencies pulling down!) but it can certainly be helpful when judging saturation, which this plug-in offers in spades.

Further handy features include Alt/Option- or right-clicking any parameter to toggle between its current and previous states, and the ability to inversely link the input/output levels and the Aggression and Trim controls by Shift-dragging on Aggression or Input. Also, holding down Alt/Option gives you much finer control with the mouse. All nice touches that help to make life easier.

The four EQ bands on the default GUI have gain knobs marked 0-10, the centre position (I hesitate to say 'neutral', as the curve is never perfectly flat) being 5. Not a decibel in sight, and it's an ethos borrowed from the hardware to encourage you to use your ears before your eyes. Each band also has a two-position switch.

A low shelf has 80Hz and 150Hz settings that change the curve: 80Hz seems a little more resonant, with a dip just above the turnover frequency when boosting, while 150Hz offers a smoother slope. The curves seem not to be symmetrical, with cuts on both settings smoother than the boosts (80Hz has no 'bump' when cutting, for example). Mid is a bell whose switch toggles between Flat and Full. Set to the former, it applies a very broad mid boost, centred somewhere around 200Hz (the bell shifts up to that frequency from a little lower down as you increase the gain), while the cut focuses on the 500-600 Hz region. Switching to Full, you get a 500-700 Hz boost (again shifting with gain) and a narrower 700-ish Hz cut.

The high band is a shelf and has Smooth and Sharp settings. The former has a slightly more 'scooped' curve than the latter, whose gentler slope has a more audible effect lower down the spectrum. Finally, the Air band is another shelf, but it operates higher up. Like the other bands this defaults to 5, but it's actually a boost-only band, with the neutral position at 0. This has an Air Shift switch that can be set to on or off. A boost when on lifts the sound up from about 5kHz, rising as you go higher up the spectrum (well beyond 20kHz), and with a broad but shallow dip around 3kHz. Switch to off and the dip pretty much disappears, resulting in a more linear slope.

»

» The outer two knobs, coloured red, control the saturation. In the hardware, Aggression drives the valves harder when turned clockwise — potentially very hard, sacrificing valve longevity for character. In the plug-in, this defaults to zero, and as you turn the knob clockwise the frequency balance and amount and nature of harmonic content changes. It's easy to overcook this — there's plenty of saturation on tap for more creative recording and mixing treatments, and way more than you'd ever need in a mastering context — but subtle treatments are possible too. A small Calibration control allows you to increase or tame the harmonic complexity, and the auto gain or Match facility, or the inversely linked controls help you rein in the level changes, even before you reach for EQ.

Popping Out

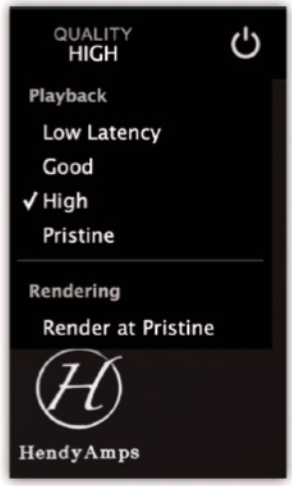
What I've written above describes various functions in isolation, and only the controls in the default GUI view. But two things makes this EQ particularly special. First, it captures the interaction between the different controls and EQ/valve amp stages. This has pros and cons, in that you can find sounds with the Michelangelo very quickly that would be hard or time-consuming to achieve with other processors, but sometimes when you try to achieve something specific you can find it a little tricky to get the balance right. Second is something that mitigates that 'con' considerably: the pop-out control panel enables you to refine the behaviour of the controls in a way you couldn't dream of with the hardware, as well as delivering considerable extra functionality.

Poptastic: in the pop-out control pane, you can pop out another window that gives you yet more control over the dynamic EQ.

With this pane open, controls beneath the Aggression knob adjust the contribution of the modelled valves: a Tube Comp knob sets the amount of valve compression (0 to 400%) and a Tube Blend knob sets the balance between triode (as on the hardware) and pentode valves, the latter sounding somewhat brighter. These make that Aggression control so much more versatile: you can control quite precisely just how softened or not transients will sound, and how smooth or brash the distortion character is. Two more knobs set the amount of crosstalk and 'spread', which determines how matched/different the modelled left and right channels are. In the default preset, there's a significant difference, which can be perceived as a lovely, subtle widening, but you can turn that off here, and you can swap the left and right models too.

Beneath the EQ controls, each band has an array of useful controls. You can adjust the centre frequency and there's generous overlap between adjacent bands. Each band is given its own Drive knob too — boosts in particular will behave differently with more drive, as they hit the valve stages harder, and you can dial things right back for a cleaner sound than the default, extending your control over the bigger saturation picture considerably. It gets better: sliders set to what degree each band acts on the Mid and Sides, or the transients and body of a sound — a wonderful facility for mastering or, say, EQ'ing an intricate picked acoustic guitar part to tame the 'boom'.

Each band also has threshold and range controls that transform it into a dynamic EQ. This doesn't negate your static EQ setting, but rather uses that as its starting point, its action indicated by a circle of virtual LEDs around the gain knob. This can expand or compress, and a neat touch is that you can invert the threshold, so that it boosts when the signal drops. A Shift-click links the threshold and range knobs of all bands. A triangle above brings up



To preserve CPU without sacrificing quality, you can set the plug-in to run at different quality settings during playback, yet still render offline at the highest quality.

a small pop-out window where you can set the duration of and sensitivity to transients, and specify the attack and release times. I can't stress just how useful it is to have access this sort of facility while you're making your broad-brush moves above: if a boost does nice things but also raises something annoying, it's easy to attend to that side-effect.

Finally, high- and low-pass filters (1-700 Hz and 1-30 kHz, respectively) each have a choice of slopes, and a conventional digital EQ has two bands with frequency (20Hz to 20kHz) and gain (±12dB) knobs. There's no Q setting, but you can set each band as a high or low shelf, or a wide or narrow bell. These have the same transient, M-S and dynamic EQ controls as the others, and while they're 'vanilla' EQs, note that they still feed the modelled output valve circuitry, which will respond to changes in signal level.

Chiselled Features

Michelangelo, the great sculptor, reportedly said of his statue of David that he "saw the angel in the marble and carved until I set him free". To my mind, that describes the way Tone Projects have approached modelling this wonderful hardware EQ: they've not only captured its essential beauty in what I have to say is a stunning-sounding model but, in delivering all the thoughtful extras, they've also revealed to us a vision of what that device might have been, were it not for the inherent limitations faced by all those who design hardware. Meanwhile, the GUI has been skilfully conceived to make the user experience simple, despite the underlying complexity of this superb tone-shaping tool — we users might think of it as a better chisel with which to sculpt our mixes and masters! In short, the Michelangelo plug-in is an equaliser like no other: invest a little time to learning how to get the best from it and it could very quickly become your go-to 'vibe EQ'.

$ $249
W https://www.toneprojects.com

ARC STUDIO

MAKE ANY STUDIO MONITORS SOUND BETTER

For the first time in the recording industry, an affordable hardware processor can be used as a correction and calibration system for any studio – home, project, or professional – without the limitations of a software-only system. ARC Studio is a unique, out-of-the-box, acoustic room correction system that instantly upgrades traditional studio monitors with breathtaking results. Finally there's a high-definition monitoring solution for all.

ARC 4 - Adjustable correction range

ARC 4 - Target curves

LEARN MORE
www.ikmultimedia.com/arcstudio

IK MULTIMEDIA
SOUND BETTER

Arturia AstroLab

Stage Keyboard

AstroLab puts Arturia's considerable collection of classic instruments into a single stage keyboard.

GORDON REID

Ever since the earliest attempts to use DSPs to emulate analogue synthesis, people have dreamed of a keyboard that can host accurate emulations of the keyboard instruments that have underpinned popular music from the 1960s to the present day. A few manufacturers have even tried to build one, but it's fair to say that none of their attempts was a commercial triumph. But perhaps that's about to change because I have in front of me a pre-release version of the AstroLab, which promises to make the sounds of Arturia's software instruments available in a single, compact stage keyboard. So, is 2024 the year when we'll be able to carry a grand piano, a Hammond B3, a Moog Modular, a Synclavier, a Fairlight and around 30 other instruments onto stage under one arm? Let's find out.

Introducing The AstroLab

The 61-key, velocity- and aftertouch-sensitive AstroLab is clearly from the same company as instruments such as the KeyLab series, but it looks slicker and smarter, and I particularly like

the ‘wraparound’ cheek design. Weighing in at 10kg, it isn’t heavy but, because it’s so compact, it feels reassuringly solid and robust. Unfortunately, one decision that was made to reduce its size was a bad one: the pitch-bend and modulation wheels are placed behind the keyboard rather than to the left of it. To me this decision is incomprehensible because the AstroLab has been designed for the stage and, if you need to place it below another keyboard or shelf in your rig, you may be unable to reach the wheels. At the very least, your wrist could be uncomfortably bent backward as you attempt to use them.

A second surprise was the positioning of the control knobs — which we’ll discuss later — to the right of the top panel. Most players are right-handed, so it would make more sense for these to be placed on the left so that they can be tweaked more easily while playing. Arturia’s own pre-release video shows the presenter reaching uncomfortably over his playing hand to demonstrate the use of these. It looks awkward, and it can’t be conducive to a good performance.

Despite its piano-shaped keys, the AstroLab has a semi-weighted synth-action keybed that Arturia claim is designed “to hit a sweet spot between pianists who expect some resistance and synth/organ players who want to be able to move fast”. This is a laudable target (even if it unintentionally insults pianists) especially in a keyboard that seeks to emulate such a wide range of instruments. But inevitably, a compromise risks pleasing no-one, so I recommend that you test it for yourself. If a hammer-action model later appears, the question will then be (as always) whether you want to risk organ swipes on a piano-style keybed or attempt to play grand pianos on a synth-style keybed.

A large encoder and its associated navigation buttons dominate the centre of the panel. Arturia have made a brave decision here, embedding the instrument’s 320-pixel display in the centre of the encoder. I can’t see any advantages to this but, as long as there are no long-term reliability issues, neither is it a problem; all is fine provided that you keep your hand out of the way while rotating the outer ring to change values or when pressing the screen down as the equivalent of an Enter button.

There are five ways that you can connect the AstroLab to the outside world. If you’re using Wi-Fi, you have two options: you can connect it to an existing network, or you can create a one-to-one relationship with your computer by making the synth a Wi-Fi hotspot. While you might choose a network for flexibility, I would recommend using hotspot mode if you’re going to connect the AstroLab to anything when performing; you never know what might happen with a public network. But bear in mind that the AstroLab doesn’t support MIDI over Wi-Fi, so you’ll need to use a 5-pin or USB cable if you want it to talk to other hardware or use it as a controller. The fifth method is to use Bluetooth. Once paired with your computer, tablet or phone, you can stream audio of up to 48kHz sample rate through the AstroLab, and the manual suggests that this is for “playing along [...] with songs that reside on your phone or computer”. It’s simple to set up and it works.

Arturia AstroLab
$1599

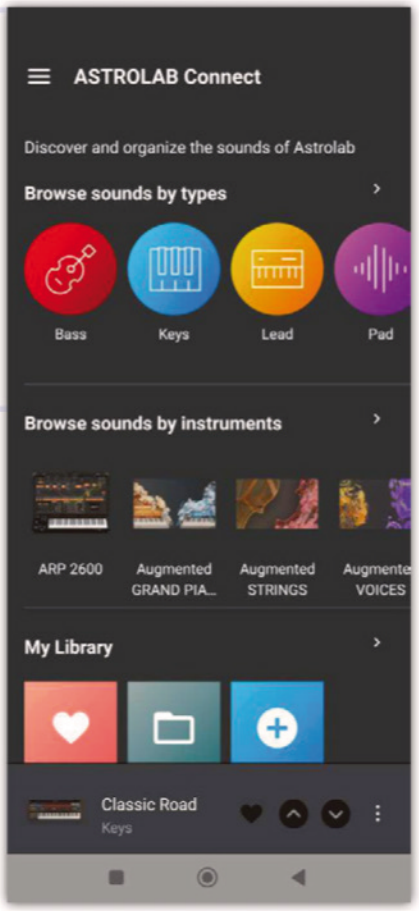
PROS
- It allows you to take a host of venerable synths and keyboards on stage without the cost, weight or hassle.
- Like the instruments on which it’s based, it can often sound remarkable.
- You can program myriad sounds for it using Analog Lab and Arturia’s software instruments.
- If you don’t want to program, Arturia’s Preset library now exceeds 10,000 sounds (although you will have to pay for many of these).
- The internal memory is large enough for any reasonable requirement.

CONS
- At the time of writing, it’s a work in progress.
- The wheels and knobs are in the wrong places.
- Some players will find its bi-timbrality constraining.
- An external power supply.

SUMMARY
The promise of the AstroLab is self-evident. If you want to take classic keyboards on stage without the size, weight and hassle of the originals (let alone the cost) it has much to commend it, and it can produce many modern sounds too. As technology advances, this could prove to be the start of a very interesting product dynasty.

AstroLab Connect

If you choose to connect the AstroLab to an iOS or Android device, you can take advantage of a dedicated librarian called AstroLab Connect. This allows you to control aspects of the keyboard from a larger display and touchscreen. Just one word of warning if you’re using older Apple products — I couldn’t install AstroLab Connect on my iPad Air (which I still use at every gig for mixing 48 channels of audio for my foldback) because the software requires iOS 13 or later, and my hardware won’t upgrade beyond the final revision of iOS 12.



Sounds, Sounds, Sounds

AstroLab sounds are organised into four levels. The first is a sound (which, to avoid ambiguity, I’ll call a patch) created using one of Arturia’s instruments within their Analog Lab software. This can be a patch supplied by Arturia or, if you have the appropriate instruments, programmed yourself. Once saved, it can (with a few exceptions and caveats) be transferred to the AstroLab, stored, and then played whether the computer remains connected or not.

Either one or two patches comprise a Preset. A Preset with one Part is called a Single and a Preset with two Parts »

» is called a Multi, which, given that this term has existed elsewhere for decades with a different meaning, is misleading; I wish that Arturia had called it a Duo or something equivalent. When two patches are used in a bi-timbral Preset, they're called Parts and can be arranged as either a split or layer with each having its own MIDI channel, octave, transposition, pan and volume settings. You can determine whether a Part responds to the wheels and pedals, and you should be able to choose whether aftertouch affects neither, one or both. But, for the moment, this isn't possible. (It always affects both.) Happily, there are no limitations on which patches you can use to populate the Parts in a Multi. This isn't trivial. If, in the past, you wanted to go on stage and play a Modular Moog with one hand and a Synclavier with the other, you tended to need a Modular Moog and a Synclavier!

Moving up to the next level, you can compile up to 128 Presets into a Song. You can then use buttons 0-9 to select the first 10 of these, but you're going to have to use the navigation system to access the rest, which means that the Song structure may not be ideally suited to your 25-minute magnum opus. The top level is then the Playlist, which contains your chosen Songs. You can also create Playlists that contain Presets without using the Song layer but, while this adds flexibility, I must admit that I would have preferred Arturia to adopt one hierarchy and then stick with it.

In addition to allowing you to play and tweak Presets, the AstroLab offers four additional sets of facilities: a single-track looper, an arpeggiator, chord and scale modes, plus four effects slots followed by a master EQ. The first of these is a MIDI recorder capable of replaying a performance once or in a continuous loop. It records velocity and aftertouch but, as yet, not the data generated by adjusting the knobs. The manual promises quantisation, swing

Polyphony

This list shows the maximum polyphony of each instrument, although the use of complex patches can cause some instruments to drop below the quoted figures. Pigments and the Augmented instruments are especially hungry and, while Arturia claim that any factory Presets based upon the Augmented instruments will have a minimum polyphony of four, the company don't quote a maximum for these. I found two problems here. Firstly, as I expected, eight notes are woefully inadequate to play many of the DX7's classic sounds. Secondly, some patches on the Matrix-12, OP-Xa and SQ-80 can exhibit an incorrect response when the sustain pedal is held and you play beyond the maximum number of notes. Having not noticed this elsewhere, I dived in to determine the problem (it turned out that the contours were not retriggering from their starts) and tested a similar patch in SQ-80 V. This was fine. I then tested it within Analog Lab and the problem reappeared. There's something strange going on here, and this is something that Arturia need to investigate.

ARP 2600 (16)
B3 (48)
Buchla Easel (1)
Clavinet (48)
CMI (16)
CS-80 (16)
CZ (8)
DX7 (8)
Emulator (8)
Farfisa (48)
Jun-6 (8)
Jup-8 (8)
MS20 (1)
Matrix-12 (12)
Mini (16)
Modular (8)
OP-Xa (8)
Piano (48)
Prophet-5 (16)
Prophet-VS (16)
SEM (16)
Solina (16)
SQ-80 (8)
Stage-73 (48)
Synclavier (16)
Synthi (1)
Vocoder (8)
Vox Continental (48)
Wurli (48)
Pigments (8)
Augmented Piano*
Augmented Strings*
Augmented Voices*

The AstroLab measures 935 x 327 x 99mm and weighs in at 10kg.

and a fixed-length record mode, but none of these were available in the review unit. Even once they've been added, there will be no overdubbing or editing functions. If you like what you've played, you'll be able to transfer your recording to your DAW and then edit it but, having done so, the only way to use it will be via MIDI because you can't return the results to the AstroLab. You can, however, store up to 127 unmolested recordings within the AstroLab itself.

The monophonic arpeggiator (which you can apply to Part 1, Part 2 or both) offers seven modes, a maximum five-octave range, and hold. It lies after the looper in the sound generation chain, which means that you can arpeggiate recordings without chewing through the memory that storing all the generated notes would require.

Chord mode can also act upon Part 1, Part 2 or both. You can select a named chord from the menus or play a selection of notes while pressing the Chord button to memorise the chord that you want. Arturia's pre-release documentation refers to parameters for strum amount and humanising the dynamics of chords, but these don't appear on the review unit, nor are they mentioned in the manual. Hopefully, they'll appear in a later revision of the firmware because they would be useful. Related to Chord mode, Scale mode allows you to determine a root note and scale, following which everything that you play is constrained by this. Arturia claim that Scale mode makes it "effectively impossible to hit a wrong note", but I find it deeply disturbing when a keyboard refuses »

» to play the correct pitches on the black notes when you select a C Major scale, or outputs an F triad when you play an F# triad.

The first two of the effects slots, FX A and FX B, host assignable 'insert' effects selected from a list of 12 types. The third and fourth are dedicated master effects: delay and reverb. When a Preset comprises a single Part, FX A and FX B are placed in series and their output can be routed through the master effects and then the EQ. When the Preset has two Parts, you can allocate the insert effects before routing their outputs through the master effects and the EQ.

In principle, the looper, arpeggiator and any appropriate effects can be synchronised on a per-Preset basis to an internal master clock or received MIDI Clock. However, there are currently some issues with this, and Arturia have confirmed that their team is working on these as part of the next update.

In Use

The philosophy of the AstroLab is that you don't need to program your sounds in detail when using it — this has already been done for you. All you need to do is find the Presets you want, organise them in useful ways, and play. Nonetheless, you can modify a Preset in a limited fashion using the control knobs to adjust any parameters assigned to them when it was created. These knobs are split into two groups of four — Instrument and Effects. In the first, Brightness controls things such as the low-pass filters on analogue synth emulations and the upper drawbars on organs, Timbre modifies other tonal qualities such as filter resonance, Time is generally directed toward contours, and Movement affects modulation. In a Multi, these knobs can affect just one Part or both, which means that you can (for example) add vibrato to one Part while leaving the other unaffected. A Shift mode also allows them to control each Part's volume and the master EQ. The second group controls the effects processors. Its first two knobs affect the wet/dry mix of the insert effects, while the third and fourth control the levels sent to the delay and reverb respectively. These again offer Shift functions, this time controlling the intensity of the insert effects as well as the delay time and reverb decay/size. The manual says that the knobs should generate MIDI CCs when turned, while a product briefing document states that they should generate NRPNs, but my MIDI analyser showed that nothing was being transmitted. I checked with Arturia, who told me that the knobs will transmit standard CCs over MIDI, while a higher-resolution protocol will be used to communicate with Analog Lab. Nevertheless, my findings were correct; none of this is implemented yet.

If you want to dive in further, the possibilities offered by the combination of Analog Lab, Arturia's software instruments and the AstroLab are enormous, although Acid V, MiniFreak V, CP-70V, Augmented Woodwinds, Augmented Brass and the latest Mini V4 and Wurli V3 are yet to be included. They're due some time in the coming months. And despite what you might read elsewhere, Mellotron V is not supported. In its place there's a subset of eight-voice (no pun intended) Mellotron patches that have been created using a new sample library, and you'll find these in a special Sampler instrument category.

Despite the omission of the latest Arturia goodies, I spent many happy hours creating patches in Analog Lab before saving them in the AstroLab's memory and combining them in Presets in ways that crumbly old proggers enjoy; a Hammond to the left and a Minimoog to the right, a Solina layered with a Rhodes, a Modular bass patch to the left and a Mellotron to the right, a Prophet to the left and a Synclavier to the right... There are approximately 2000 such combinations available, so it's almost as if one has the opportunity to play the world's largest keyboard rig. And how does it sound? Lovely! I can forgive a great deal for the opportunity to have a single keyboard that sounds all but indistinguishable from my large, heavy, fragile and sometimes unreliable vintage keyboards. And that's the key to the AstroLab. While you could use it as your stage piano or organ emulator, I envisage it sitting above a workstation or one of the more common stage keyboards where — barring some exotic requirements — it could add the sounds of almost anything else you might need. The AstroLab's memory is a healthy 22.59GB, with 9.43GB used to store the factory Presets and their associated samples. If I've calculated this correctly, the remaining space is enough to hold around half a million additional patches if you don't save any more samples!

There is, however, a caveat. Due to processing constraints, the convolution reverbs used within the Augmented

The Rear Panel

Starting at the left of the rear panel, you'll find conventional 5-pin MIDI in and out sockets but no thru. Instead, a thru mode re-transmits incoming MIDI (whether received via 5-pin or USB) through the out socket, mixing it with whatever you generate on the keyboard itself. To the right of these lie four quarter-inch analogue control inputs: programmable expression and sustain pedal inputs plus two auxiliaries. Next come stereo XLR/TRS mic/guitar/line audio inputs and their associated gain control. If you use XLR plugs, the signal passes through mic preamps. It would have been nice to be able to obtain a tad more gain from these, but I was still able to perform my 'Mr Blue Sky' impersonations with no problems. Alongside these you'll find balanced, quarter-inch audio outputs and a stereo headphone output which would, of course, be better placed at the front of the instrument. There are two USB sockets: USB 2 Type-A for connecting external storage devices, and USB 3 Type-C for computer connection. The final socket is for the external 12V/3A power supply, and this offers a collar on to which a retainer screws. This is a much more robust connection than you get with most external PSUs, but I would still have preferred an internal power supply and an IEC socket, not least because, if you trip on the AstroLab's power cable, you might find yourself snapping it or pulling the keyboard off its stand.

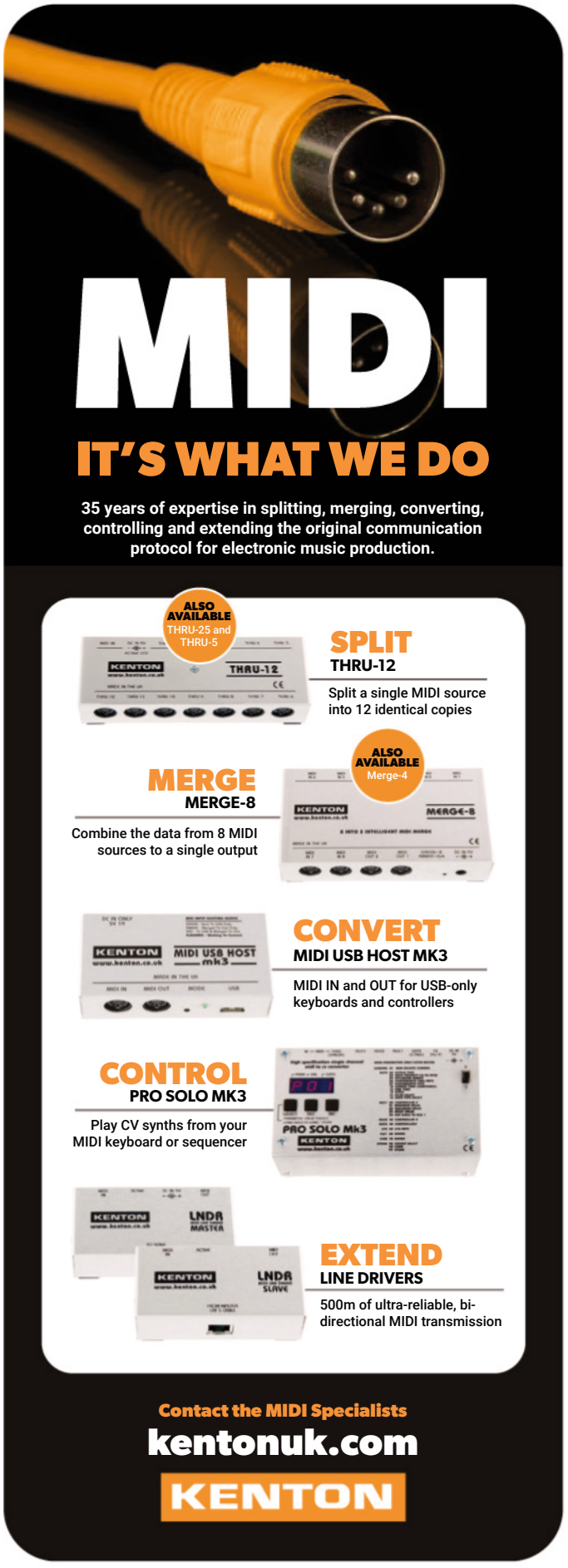
series, Solina, B3, Farfisa, Stage-73, Clavinet and Piano are discarded, and it appears that the monophonic instruments and others offering 48-voice polyphony also lose their integrated reverbs. This means that you have to use Analog Lab's effects to replace them, which might modify the sounds of some existing patches. I can't claim that this caused me any grief but, if you've used the original software instruments on your album, you might want to check for differences when building your Playlists for the world tour.

Inevitably, there are other niggles. For example, you can't copy a Playlist directly from Analog Lab to the AstroLab; for the moment, you have to use a USB memory stick, which rather breaks the philosophy of the marriage of the two products. More serious is the delay when selecting a Preset that uses extended samples. This can take several seconds and, while the AstroLab allows you to hold existing notes as a new Preset loads, you'll need to take the lag into account when creating your Playlists. Another oddity was that the review model continually spewed out a stream of MIDI pitch-bend messages. Waggling the wheel could make this stop for a while, but it restarted a few moments later. I encountered several more issues as I delved deeper, and discovered some further differences between the keyboard and its manual, but these all boiled down to the pre-release firmware so I won't belabour the point. However, you should be aware that the next revision isn't due until after the product launch, so you may need to make allowances if you get your hands on an AstroLab immediately following its release.

Final Thoughts (For Now)

Ignoring the fact that this review was performed on an instrument hosting unfinished firmware, the AstroLab clearly has a great deal to commend it. So, would I change anything about it? Of course I would. Most obviously, it would benefit from being wider. A semi-weighted 76-note version would be much more useful (and playable) when using split Presets, and a hammer-action 88-note version will be wanted by players who intend to use it as a stage piano. Then there are the issues with the positions of the wheels and the performance knobs. But it's another item on my wish-list that will be the hardest to satisfy. Please, please, please can I have a multitimbral version that offers 16 Parts and sufficient split points to take advantage of them? Even allowing for the inevitable increase in price, I might find that to be almost irresistible and I'm sure that I wouldn't be the only one. I must admit that I'm very much looking forward to witnessing the evolution of the AstroLab.

$ $1599
W www.arturia.com



Neve 1073SPX-D

Channel Strip & USB Audio Interface

Neve's latest 1073 variant bridges the gap between microphone and computer.

Neve 1073SPX-D
$2995

PROS

- An authentic 1073 input channel with the convenience of USB interfacing.
- Easy to integrate as hardware insert.
- Retains all of the analogue functionality of the SPX variant.
- Can act as an ADAT expander, analogue monitor controller and powerful headphone amp if you don't need the USB connectivity.
- Sounds great, as you'd expect!

CONS

- Headphone monitoring arrangements not flexible enough for all use cases.
- Blend control doesn't work with the analogue monitor inputs.
- A secondary line input would be nice.

SUMMARY

The SPX-D is the first fully featured 1073 input channel that is also a plug-and-play USB audio interface.

SAM INGLIS

More than 50 years after it was introduced, the Neve 1073 remains the world's most iconic mixer channel strip. It's inspired countless imitators, and Neve's own product range now contains no fewer than 10 different products referencing this trademarked four-digit number. The 10th and newest of these is the 1073SPX-D.

Back in January 2018, Hugh Robjohns reviewed the Neve 1073SPX, a single-channel processor that includes the classic 1073 preamp and EQ circuits in a convenient 1U format. As well as the expected Marconi knobs and Mahjong-tile buttons that govern its analogue features, this sports a couple of small red buttons with associated LEDs on the top right of its front panel. At launch, these buttons were intended to control an optional digital card that would add AES3, word clock and FireWire connectivity.

In the event, however, this digital card was never released, so although the SPX continues to be a popular way of integrating a 1073 input channel into a modern studio, it can't talk directly to your computer or other digital gear. And now it never will, because that power belongs to the new 1073SPX-D. This, in a nutshell, is a 1073SPX with integrated digital connectivity, USB interfacing and monitor control.

More Than Digital

The SPX-D can be used as a purely analogue device, and in that role, it does everything the SPX can. It thus offers the 1073 preamp and EQ designs in their most fully developed incarnations, with transformers on both input and output, whilst adding modern conveniences such as a balanced insert loop that can be placed before or after the EQ, and a front-panel combi jack that overrides the rear-panel XLRs when engaged. However, the SPX-D also has analogue monitor control features not present on the SPX. These require front-panel space, so the DI input for electric guitars has been folded into the front-panel combi socket. A happy side-effect of this is that DI signals now pass through the input transformer, with the pad switch dropping the impedance from 2MΩ to 200kΩ.

The rear panel of the SPX-D is quite a bit busier than that of the SPX.

"I particularly like the the Digi button, which makes it easy to use the SPX-D as a hardware insert without repatching, regardless of whether you're hooking it up over USB or as an ADAT expander."

Like that unit, it employs an external switch-mode power supply, but this time it connects using a five-pin XLR. The SPX's quarter-inch insert sockets and XLRs for mic in, line in and line out are joined by two further pairs of XLRs labelled Monitor In and Monitor Out, while the leftmost part of the rear panel sports optical ADAT in and out sockets, a BNC word clock output and a Type B connector for USB interfacing. (Neve prefer this to the current Type C standard for reasons of robustness and long-term reliability.)

Returning to the front panel, the SPX-D inherits the SPX's output attenuator to allow the input side of the unit to be driven harder without overloading its A-D converters or downstream devices connected to its analogue outputs. This has a push action that cycles the seven-segment LED meter between three different points in the signal chain. On the SPX-D, however, it's joined by two further knobs, which have their own push actions.

Monitor Control

The most important of the added knobs is the rightmost one labelled HP/LS Level. As that suggests, it simultaneously governs the volume of the front-panel headphone socket and the rear-panel stereo XLR monitor outputs. A brief press on the button mutes the XLRs whilst leaving the headphones active, but there's no way to set their levels independently; this is a shame, especially given that Neve's more affordable 88M interface does have a separate headphone level control. The HP/LS Level knob has a detent at the centre position, allowing you to return easily to a preset monitoring level. However, the headphone amp is seriously pokey, and with modern low-impedance headphones, the detented position is way too loud for comfortably listening to mastered material.

A longer press on the HP/LS Level pot cycles through four different source options. In the first, the SPX-D's Monitor Out XLRs and headphone output simply present the input signal, panned centrally. In the second, they present one of two stereo playback returns from the USB interface, so this is the mode you'd use if you want to monitor inputs through the DAW. The third mode presents a balance »

ADAT Expander Mode

If you want to take advantage of the 1073SPX-D's digital connectivity but you already have another audio interface, it can be used as an ADAT expander. It's not necessary to put it into a special mode to do this; you just don't connect the USB cable. The mic or line signal coming into the 1073 is presented on ADAT out channel 1, or split across channels 1+2 at high sample rates (the highest rate supported in expander mode is 96kHz). The digital signal reaching the ADAT in, meanwhile, is treated like the DAW return in USB mode. This means you can use the Blend control to achieve a suitable monitor balance between the live input and the signal arriving at ADAT 1+2 or 3+4, and also that a signal coming in on ADAT channel 3 can be substituted for the 1073's line input and processed with its EQ.

The front-panel Sync button is used to put the SPX-D into internal clock mode, in which case the downstream device needs to be set to clock to it. Disengage Sync and the SPX-D will try to follow an incoming ADAT clock, but unfortunately there is no visual feedback to indicate successful clocking or the presence of a digital input signal.

As well as its digital connectivity, the SPX-D has stereo monitor inputs and outputs on XLRs along with a dedicated line output from the preamp/EQ.

The Digi button at the lower left of the front panel patches DAW return or ADAT channel 3 into the line input, making it easy to integrate the SPX-D as a hardware insert.

» between input and playback signals, which is determined by the middle of the three knobs, and it's this mode you'd use if you want to audition inputs with zero latency whilst also hearing the backing track from the DAW. A fourth mode switches the monitor source to the rear-panel Monitor In XLRs, for situations where you want to employ the 1073SPX-D as a monitor controller in association with a different soundcard or audio interface. This setting, alas, doesn't allow the input signal to be blended into the monitor path.

The top right area of the panel inherits the same two tiny red buttons and strip of LEDs that appear on the SPX, but they are now more than decorative. Amber LEDs indicate the current sample rate, and the leftmost button cycles through these in situations where the SPX-D is being used as a standalone digital source (see box). A blue LED indicates successful USB connection, and the Sync button toggles between internal and external clock signals where that is appropriate.

Over USB

Like Neve's 88M interface, the 1073SPX-D is class-compliant and thus requires no driver installation on macOS. Nor is there any control panel utility, as everything is handled either from the front panel or the Audio MIDI Setup utility. Windows users will, as usual, need to install an ASIO driver to work with most DAW programs.

At base sample rates, the SPX-D presents 10 inputs to your DAW and 12 outputs, the last eight of each being the ADAT channels. The preamp/EQ signal appears on input 1, while input 2 is unused. This is a shame, because it's not hard to imagine applications for a second input. For instance, you might hope to be able to use the insert return as a separate input, so that a line-level signal could pass through the EQ and into DAW input 2 whilst a mic is being recorded through the preamp and DAW input 1. A second line input would also allow an SPX-D to be paired with an SPX for stereo recording; as it is, stereo requires two SPX-Ds (which, on the plus side, could be configured as a stereo hardware insert).

The 1073SPX-D has more virtual than physical outputs, since your DAW can address two stereo pairs in addition to the ADAT outs. This might not seem all that useful at first, because it's only ever possible to monitor one or other of these output pairs, but the reason for it becomes apparent when you engage the front-panel Digi button. Present but non-operational on the SPX, this comes into its own on the SPX-D, and allows the analogue line input to be replaced by DAW return 3. The idea is to make the preamp and EQ easily accessible as hardware inserts within your DAW, and it works very well. Apart from that, the 1073SPX-D is pretty much 'plug and play'.

To D Or Not To D?

The Neve 1073 remains probably the first choice of input channel for many, if not most engineers and producers. But prior to the advent of this SPX-D variant, there hasn't been a 'full fat' version with EQ and insert points that could be hooked up directly to a computer and used without additional hardware. That's clearly the niche that Neve are aiming to fill here, so have they managed to make the SPX-D do everything you need, or would you be better off buying an SPX along with a third-party audio interface?

That, I think, very much depends on your intended use. These days, it's not uncommon for artists to take a compact but high-spec recording rig on the road with them, to track vocals in hotel rooms or tour busses. In many ways, the SPX-D is the ideal product for this role, and its insert points make it trivial to patch in the ubiquitous Tube-Tech CL1B (other compressors are available, as they say on the BBC). The fly in the ointment is that it has only a single headphone output: fine if you're self-recording, not so much if there's an engineer to cater for as well. Granted, the headphone amp should be capable of driving two pairs of cans through an analogue splitter, but when a session really matters, you don't really want to be relying on kludges. Generating separate headphone feeds for artist and engineer will require a second amp fed from the monitor output, and even then, both will carry the same mix and be governed by the shared volume control.

The analogue monitor inputs, meanwhile, are an interesting addition, but they don't really have much of a role to play if you're using the 1073SPX-D as a USB interface or ADAT expander — and if you're not, they probably don't justify the extra cost over the plain old SPX on their own, given that it's not possible to blend the input signal and monitor signal at the headphone amp. Finally, I imagine that some users would still prefer the AES3 digital I/O that was planned for the SPX digital card to the ADAT Lightpipe protocol that Neve have chosen here, although that works perfectly well.

But if there are areas where the SPX-D isn't quite as flexible as you'd hope, there are others where it exceeds expectations. I particularly like the the Digi button, which makes it easy to use the SPX-D as a hardware insert without repatching, regardless of whether you're hooking it up over USB or as an ADAT expander. And if you don't need the built-in monitoring, you can repurpose the monitor outputs as mults to distribute the input signal to a redundant recorder or another processor. Most of all, the SPX-D remains a fully featured 1073 channel strip, with input and output transformers, an output 'fader' and all the other features you'd expect on the input side. This is not the case with the 1073OPX, the only other 1073 model that currently has USB interfacing available as an option. If you want an authentic, complete 1073 input channel that can deliver its audio goodness straight into your Mac or PC, the 1073SPX-D is currently the only game in town. It's a 'proper' 1073 with a USB socket — and it sounds great.

$ $2995.
T AMS Neve +44 (0)1282 457011
E info@ams-neve.com
W www.ams-neve.com

Ableton Live 12
Out now

Hologram Electronics Chroma Console

Multi-effects Pedal

Whether it's sitting on your guitar pedalboard or nestled between your synthesizers, this weird and wonderful effects box oozes vintage character.

ROBIN VINCENT

Something about the styling of Hologram's Chroma Console pulls you in. The wedge shape is jaunty, the off-white enclosure smells of old gear and the primary colours remind me of stripes and logos of the 1970s. It triggers all those nostalgia neurons that get us misty-eyed and appreciative of the simple things. The Chroma Console is a multi-effects unit that's designed to drive movement, eccentricity, grit and vintage instability into your sound. And although it is, technically, a 'guitar pedal', Hologram have opened it up to a wider variety of uses, so for this review I'll be plugging in both a guitar and a bunch of electronic instruments.

Colour-infused Multi-effector

You could see the Chroma Console as a compact pedalboard, a versatile multi-effects box or the shiny thing you put on the end of your mix. It has obvious controls, great visualisation and a sense that you know exactly where you are. (This is not how I felt with Hologram's previous stompbox effect, the baffling but beautiful granular and micro-looping Microcosm, reviewed by Simon Small in *SOS* September 2022: https://sosm.ag/hologram-microcosm). Here, the navigational clarity is excellent. It's like they listened to all my frustrated Microcosm murmurings and actively set out to design an interface that even an idiot like me could grasp. Bravo, I say.

Inside the box, 20 effects are organised into four flavours or 'modules' and, running five effects each, these modules offer a multi-layer and multi-focus road to multi-effect happiness. The Character module contains drives, preamps and fuzzboxes. Movement offers modulation and pitch-shifting. Diffusion brings in some flavours of delay and reverb. Finally, Texture provides the overriding vintage vibe. Choose an effect for each flavour, tweak it with a couple of knobs, and you have the chewable sound of distressed gear all over your audio.

The first three modules have two knobs; the last one has a single knob with a global wet/dry mix knob above it. The button beneath the knobs is used to select one of the five effects and lights up colourfully to reflect your choice. You step through them in turn, so you hear each effect along the way to the one you want. That may not be ideal for some users, but the intention appears to be that you build your sound with one effect from each module, rather than trying to move between effects within a module. The knobs control the main parameters, of which there are but a few. Each knob has a secondary function, accessible via some mild finger gymnastics performed on the four buttons, and these are clearly

laid out on the black strip that does a great job of keeping you from referring to the manual.

Modularity

In the Character module, we have: the tube-like Drive; Sweeten, which adds EQ, compression and saturation to a preamp; the vintage-voiced Fuzz, which then gets a resonant filter with Howl; and finally, Swell, which is a tricky-to-master envelope-triggered volume swell. The top knob controls the Tilt brightness or, its secondary function, fine-tunes the headroom for more or less distortion.

The tone and responsiveness of Character was superb on guitar, and I felt I could really lean into it. It also did a good job of beefing up keyboard instruments and anything with a bit of dynamics. Drive and Sweeten were particularly thumping on drums, assuming you like things chunky. Away from the guitar, Swell is a bit more hit-and-miss, as I could never seem to get enough level to it from a synth without going overboard.

It's in the Movement module that the effects begin. There's a Doubler stereo double-tracking effect. Vibrato gives a nice bit of pitch wobble, while Phaser is that classic swirling ride on a fairground waltzer. Tremolo chops and flutters, and Pitch shifts you up or down up to an octave, with a slightly disconcerting delay. You have control over the Rate and Amount, but all the real magic happens with the secondary level Drift control, which dials in a sense of vintage decrepitude: it injects occasional momentary pitch-shifts into the Doubler, it pours gooey tape instability onto the Vibrato, it messes with the Phaser waveform, distresses the Tremolo and wrecks the Pitch shifting, even if you haven't added any.

The Diffusion module is the ambient playground of delay, reverb and general weirdness that Hologram are famous for. Within it, Cascade is a bucket-brigade delay with some enjoyably ridable self-oscillation that can quickly get out of hand. Reels is a nicely worn-out tape echo. Space blends between five reverbs, to give a great range of size and depth. Proper weirdness is found in Collage, a looping delay that seems to happen spontaneously. Playing with the Time knob sets Collage into spasms of granular and back into looped phrases, teasing you by pretending to be consistent before going off and doing something else. Reverse is a tape running backwards, turning what you play inside out. It pitch-shifts in a similar way to the Pitch effect, which gets really creepy if you mix the dry sound back in.

Drift plays a large part here too. It degrades and disintegrates the repeats from Cascade and Reels. In Space and Reverse, it adds a touch of vibrato, and in Collage it flips the pitch and speed all over the place, into bizarre and haunting occurrences.

Finally, Texture is simpler, more of an overall disturber of your sound, and it could feasibly sit on a mix bus or at least at the end of a chain. Filter is, by default, a tilt filter but you can also set it up as a low- or high-pass filter, with the Amount knob setting the cutoff frequency. Squash is a compressor with overdrive at the extremes, and is the sort of thing you could leave in as a default for 'glue'. Then it gets more deliberately interesting, starting with Cassette. This evokes memories of Portastudios that our nostalgia-riddled minds like to call 'treasured': beautiful when used subtly and fabulously dodgy when pushed. Broken removes the warm treacle of Cassette to leave in the worn-out mechanisms of mangled machines. Lastly, we have Interference, which introduces all kinds of glitches inspired by telecoms networks and radio static.

Pulling It All Together

Individually, each effect and each module has a lot of scope for exploration and enjoyment, but the Chroma Console wants you to chain the four modules into a definitive patch. The front-panel order of things feels very natural, but if you want Space to push a cosmic reverb into the Howl before being Broken and emerging in a Tremolo, you can re-route the modules in any order you wish.

This all throws up some challenges of how you manage the four modules with your feet, especially as they don't have individual bypass footswitches. »

Each module can host one of five effects, and though they run left to right by default, the modules can be placed in any order.

Hologram Electronics Chroma Console

$399

PROS

- Sounds gorgeous.
- Thick vintage vibes.
- Compact pedalboard replacement.
- Easy to use.
- Gesture recording keeps things lively.
- Built-in looper.

CONS

- Too simple for many.
- Clunky preset selection.
- Easy to overuse.
- You're stuck with Hologram's choices.
- Could do with some rubber feet.

SUMMARY

The Chroma Console is a compact pedalboard of colour and character that swims in a lovely warm vintage soup. It might be too simple and too gooey for some but heaven for the rest of us.

» The pedal can store up to 80 presets, which store everything including effects routing, primary controls, gestures and tempo, but I've found this to be the least enjoyable part of the pedal. Being able to store presets is great, but the clunkiness of storing and retrieving them not so much. It's not that it's difficult, but that it feels too laborious to work well in live performance. You have to hold the Bypass button to enter Preset mode, and then you can shift up and down presets with the two footswitches and then long-hold to exit with your new preset loaded. That amount of footwork does not lend itself to switching presets in the middle of a song.

Having said that, there's a clever way around this. It's a bit of a compromise, but the more I use it, the more I believe it to be completely fine in the majority of cases. The pedal has a Dual Bypass function, meaning you can set the bypass switch to turn off a selection of modules rather than all of them. That way, you can have your Drive on and drop the Reels in and out, or bring in the Tremolo and Interference, or have everything going and then bypass Diffusion and Texture. Of course, individual bypass controls would be great, but that would make for a much wider pedal. I think what Hologram have done here is enough to mean you don't have to use the preset system to switch effects while you're playing.

As well as MIDI I/O, there's an expression pedal input that can be mapped to any control.

In Use: Guitars & Synths

Playing guitar through the Chroma Console feels very normal and natural, for at least the first couple of modules: Driving into a Doubler, or Fuzzing into a Phaser is just another day at the office. As a compact multi-effect it has a good tone, feels warm and full and doesn't give me too many things to distract me from my playing. Once you engage the Drift, things start getting different. The tug of nostalgia is palpable as you surf through those magnetic vibes. Then, as you push into Diffusion, it's less of an ambient playground and more of a sticky feeling of festivals.

With synths and other keyboards I jumped straight in with everything maxed out, and it felt like something out of a dream. As I played the piano, I could be wandering through an art-house cinema, warbling through deliciously trippy environments or listening to faded recordings I made 40 years ago. I was drawn into the Texture side, swapping between Cassette, Broken and Interference because it made everything haunting and gooey. With the time-based effects, it was the Drift control that got most of my attention. It pushed the sound off-kilter, tripping up in time and pitch while the music fell apart.

Conclusions

I've had a thoroughly good time with the Chroma Console. I could see it being my everyday guitar pedal or knocking around my synths and modular, ready to dip my music in sumptuous tape-style saturation and instability. There was lots of room for knob-twiddling and enjoying changing the effects as part of my instrument. The gesture recording gives it a lot of scope for interesting animation and the single loop capture is a nice hidden extra. But it's also seriously limited in what you can change. Hologram have made a bunch of assumptions over how these effects should go, and if you prefer having 14 knobs to sculpt a single effect, then this is definitely not for you.

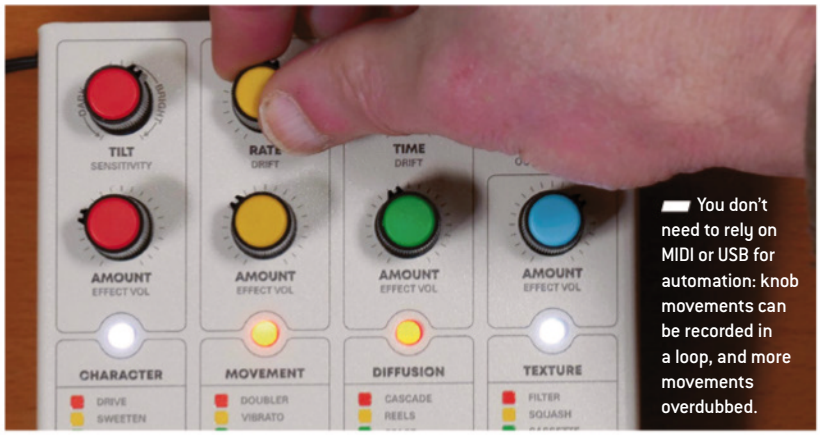
The box is simple and easy to use, almost to a fault. The sound is delicious, like pouring honey over your cables, but it's also easy to overdo. There are times when you'd like to use two effects within one module: if I could put the Drive into the Swell and the Reels into Space, I'd be a very happy man. But the key to enjoying the Chroma Console is embracing Hologram's curation and sinking into the sheer gooiness of it all. ■■■

$ $399.
W www.hologramelectronics.com

Automation, Gestures & Capture

You can automate parameters over MIDI or USB, and the expression pedal input can be mapped to any control you like. But automation is also baked into the box. With the press of a couple of buttons, you can enter gesture recording mode, which records the movements of the primary knobs. It's like putting in a manual LFO or throwing in some crazy accent, speed change or a slow plunging in depth. The gestures loop, and can be stacked up with whatever knobs you want to move.

Another big feature is hidden behind the Tap switch. If you hold Tap down, you can capture up to 30 seconds of recorded audio (pre or post effects) that will then loop indefinitely. There's no overdubbing, just a single loop that gets replaced if you do it again, but there's a sustainer variant whereby if you hold for a very short time the audio will be captured and soft-faded into endless and seamless pads. Except, I could never get it to do that — it would grab a short piece and then stutter it out like a ratchet. It was softer around the edges than longer captures but a long way from being a seamless pad. Maybe it's all in the expectation.



You don't need to rely on MIDI or USB for automation: knob movements can be recorded in a loop, and more movements overdubbed.

Hear What You've Been Missing

The Nuance Select is a monitor controller designed with the belief that nothing should stand in the way of the sound that reaches your speakers. With exceptional sound quality, it sets new standards for audio reproduction, letting you hear the true, uncolored sound of your monitors with astonishing clarity. Reveal subtle details that might previously have been obscured, and enjoy a more authentic, immersive stereo field. By giving you the full picture, the Nuance Select empowers you to take control and make every mix decision with confidence.

Nuance
select
Studio Monitor Controller

Radial engineering

Hear the True Sound of your Monitors
www.radialeng.com

Minimal Audio Current

ROBIN BIGWOOD

Software Synthesizer

Minimal Audio's Current is nothing if not ambitious, and not just as an instrument...

Minimal Audio are a Minneapolis, US-based company that have carved a niche for themselves as developers of distinctive plug-ins and sound libraries. Current, their most recent and complex product, is also their first virtual instrument. But actually it's more than that: it's a flagship that brings with it the rest of the Minimal product fleet, so to speak. In investing in Current you also get a whole suite of goodies that interrelate as a kind of mini ecosystem. It's an intriguing idea, but with so many capable soft synths out there already, is it enough to make you jump ship?

High Voltage

Current is an über-synth in the style of Arturia's Pigments. It's not based on any one thing from the past, and instead almost every aspect of it is ultra-flexible and configurable. It's still essentially a subtractive design, but of the most lavish, well-equipped kind.

Constituting the oscillator part of the signal chain is a handful of sound generators running in parallel. Two wavetable oscillators encompass everything from typical analogue waveforms to complex digital and sample-like timbres. Literally hundreds of wavetables are built in, and they're all of the real-deal, multi-frame, smoothly morphable type. Two additional parameters, Wave and Warp, dial in waveform distortion with over 20 modes each, emulating the sound of oscillator hard sync, bit reduction, filtering, formant and frequency shifting, and a lot more with just the turn of a knob. Also, like Xfer Records' Serum and Vital Audio's Vital, Current can synthesize wavetables from audio you drag and drop into it, and import existing wavetable files from other synths too.

The granular oscillator here is a nice implementation of this sound-generating tech, with parameters for playback position, spray/jitter, grain rate/density (sync'able to clock), flexible grain envelope shapes, and embedded filters. Finally there's a sampler with a sub-oscillator that will do classic chipmunk-susceptible re-pitching or time-stretch your samples to keep their original duration, and more plausible formant content. There's no pitch- or velocity-driven sample switching but samples can reverse, loop and crossfade, and there's an embedded multi-mode filter

With their own tabbed interfaces the granular, sub and sample oscillators are no poor relations: they underpin many of the more experimental presets for one thing.

and (like the wavetable oscillators) a unison feature offering as many as 16 detuned voices/layers and stereo spreading. The sub-oscillator is unusually sophisticated, with parameters that redistribute the intensity of its harmonic spectra, and detune upper harmonics. All five of these oscillator types can play at once, though in practice you're more likely to use a smaller combo.

Next along the signal chain we get two filters, which can operate in series or in parallel. With identical capabilities they offer filter responses grouped into categories called Basic, Morphing, Creative, Formant, Comb and Phaser: well over 50 different types. Together with the inevitable cutoff and resonance parameters there's also Spread, which decouples left and right signal paths for stereo widening effects, and Morph, which for the Morph filter responses provides that wonderful continuous low-/band-/high-pass transition typical of the Oberheim SEM, and for others dials in complex slope/peak distortions or variable spacing.

Further expanding the harmonic treatment options is a whole bank of audio modulation paths. Any oscillator can be frequency- or amplitude-modulated by itself, by any other, by a noise source, or by the output of either filter. Audio-rate sonic shredders, fill yer boots...

Alongside, the modulation scheme is similarly open-ended. We could be here all day, so I'll summarise it. There are 10 modulation sources: one AHDSR envelope that's hard-wired to amplitude, and nine others freely assignable. They can be further envelopes, LFOs, curve generators (an LFO/envelope mix on steroids, with a variety of preset shapes), or envelope followers (tracking oscillator or filter outputs). All are superbly equipped: for example, envelopes have variable curve shapes and can be looped. LFOs are sync'able, key-triggered or free-running and morph between classic analogue waveforms.

Further modulation comes from the keyboard: key-tracking, note-on and -off velocity, pitch-bend, mod wheel and aftertouch. If you prefer an MPE controller you can switch these to become Strike value, Glide, Slide, Press and Lift instead. All with variable value limits and response curves. Four macro knobs look conventional at first but turn out not to be. They can modulate multiple destinations, and can themselves be modulated by anything else.

Assigning modulation is done in one of two ways. The first is to right-click on any parameter and choose 'Add Mod'. The second is to drag a 'token' from a modulator and drop it on top of a parameter, which will then be equipped with a little horseshoe-shaped knob that both indicates it's being modulated by something, and lets you dial in the modulation depth you want. Right-click it and you can set unipolar or bipolar response, and also choose a further modulator to control modulation depth. Then, moment to moment parameter values are shown by animated rings around knobs, and in the case of the wavetables oscilloscope-like waveform displays. There seems to be no limit on the number of parameters that can be controlled by a single modulator, but apparently individual parameters max out at three simultaneous incoming modulation signals.

All told, Current's synthesis scheme is an embarrassment of riches. Anything you can't achieve with it, in pure harmonic terms, is probably not worth doing. But as is the way in the 2020s, it's only part of the picture. Effects are arguably at least as important an element of sound design these days, and Current does not mess about in this area either.

I mentioned Minimal Audio's existing reputation for effects plug-ins, and it turns out that their entire line-up is built into Current (barring the flagship Rift distortion processor, which appears here as Polar Distortion, a sort of 'Rift-lite') and can be deployed how you like in nine effect slots. It's clear from only a moment's experimentation that all are of top quality and are very controllable, for treatments from subtle to cataclysmic, often with surprisingly few controls. Effects parameters are fully available to the modulation scheme too, so can be animated as part of synth presets. Taking staples like the Swarm reverb and Cluster delay as a case in point, I appreciated the fact that they can quickly »

Minimal Audio Current
$199

PROS
- Impressive programming depth, with deep, highly configurable oscillators, filters and modulation.
- Equally impressive clarity and economy in the user interface.
- Excellent onboard effects.
- Factory and net-accessible presets are frequently big, ballsy and full of character.

CONS
- Max eight-note polyphony per instance.
- Sound library has strong contemporary character, at the expense of simpler vintage timbres.

SUMMARY
A big, fully featured soft synth that's as strong on pure synthesis as on its integrated effects processing. It connects to the net too, accessing additional presets, wavetables and samples: the associated subscription/membership scheme even gets you new plug-ins for your DAW.

» get very synthetic and experimental in character, for some really out-there sounds. That seems appropriate given the versatile, no-holds-barred style of Current as a whole.

Plugged-in

So far, so good, but whilst Current's feature set is colossal and the implementation impressive, it's not fundamentally different to some similar synths I've already mentioned. The Stream panel is where that changes. Essentially, Current utilises an Internet connection in a way I've not really seen before in a synth.

Stream is a sort of dedicated browser and file management system, with audio previews, content filters and various different ways of sifting through the material on offer. But rather than trawling through the murky depths of your hard drives for long-forgotten WAVs, this browser connects to Minimal Audio HQ on the Internet, and draws from a large pool of content: presets, wavetables or WAV-based sounds ready to be loaded into the granular or sample players. It can alternatively see local content, but only ever Current-specific stuff.

At the time of writing, looking at the Cloud offerings, a precise total number of available presets isn't given, but it looks to be well into the hundreds and quite possibly the low thousands. Fewer wavetables were available, but still ran to some hundreds. And as for sounds, it's certainly thousands. The promise from Minimal Audio is very much for this number to increase significantly over time.

Downloading this content makes it available locally, so there shouldn't be any nasty gotchas when you take your laptop into the wilderness, or on a flight, and find that a lack of Internet access prevents you from picking up where you left off. However, like many an online subscription-leaning scheme, there are some realities to be aware of. See the 'Current Contract' box for more on that.

Bun Fight

Current clearly has a formidable synthesis and effects architecture, but how does it actually sound? In one word: complex. In a couple more: really impressive.

It's not that you can't get simple and clean results from it, and fast too. That's revealed if you initialise a preset and load just simple oscillator waveforms and conventional 12 or 24 dB filter responses. It can sound very smooth and warm indeed, especially with each wavetable oscillator capable of anything up to the 16-voice unison I mentioned before, and without affecting overall polyphony. There are silky filters, fat filters and sizzly filters, and with a touch of chorus and reverb you're already through and way beyond vintage analogue polysynth territory. As part of testing I pushed all the oscillators to ultrasonic extremes and heard little or no side-band or aliasing gunk. A global option to run the synthesis engine with 2x or 4x oversampling should do away with it completely, at the expense of some additional CPU load. »

There are no fewer than nine effects slots in Current, available to be filled (for the most part) with full, equivalent versions of Minimal Audio's existing DAW plug-in effects. Also seen here is the OP-1-alike keyboard display, and a typical effect interface, from Cluster Delay.

Tactile

The lower section of Current's window (which is continuously resizeable with a mouse drag, by the way) does double duty, sometimes showing modulator programming detail, and the rest of the time showing a keyboard display that's pretty much a lift of the layout Teenage Engineering use for their OP-1 hardware synth. It's stylish without having any other particular functionality. More important are the nearby Chord and Arp panels.

Chord generates note stacks from single key presses, and there's some real sophistication. Overall key can be set, and individual chord types specified for every degree of a scale, including the presence of bass notes, texture and density of voicing, added sevenths and ninths, suspended seconds and fourths, and a Strum option to arpeggiate the result. A menu can recall over 40 ready-rolled possibilities, and you can save your own.

The arpeggiator is good too. There are multiple patterns, range options and rhythms, and as in so many other places in Current the relatively few controls will take you from the basics to the outer limits, and support both careful programming or hit-and-hope experimentation. The fact that there aren't any polyphonic modes is mitigated by the fact that it can work together with the Chord section I just mentioned. The combo will then either arpeggiate whole chords, or feed chord info into the monophonic arpeggiator, which is very neat.

Finally in this section are some options for monophonic mode, legato triggering, and glide. And whilst glide time can be modulated, there are no other options: one of the very few areas of Current that feels at all undercooked. I'd happily take time/rate and curve options here, and also some more sophistication surrounding polyphonic glide.

Focusrite | Scarlett

WINNER
SOS AWARDS 2024

One simple step to keep you focused on your music.

www.focusrite.com

Focusrite's new **Scarlett 4th Gen** interfaces are packed with exciting and helpful innovations, including **Clip Safe**.

With **Clip Safe** engaged, you'll never lose a take. Plug in to your **Scarlett** and while you're playing, **Clip Safe** checks your levels and turns down the gain if you get close to clipping. No compression. No limiting. No effect on your sound. Stay focused on your music, without losing that perfect take.

**Scarlett 2i2 and 4i4 4th Gen only*

Learn More

The Stream view is where Current leaves behind most other soft synths. The content browser features are familiar enough, but the net-connected cloud content may well be a unique selling point. Seen here is a preset browser poised to dig into some curated preset packs. The 'Bloom' panel is an example of a search for sounds, with dedicated buttons to load these samples straight into the granular or sample oscillators.

» However, with Current, a world of altogether noisier, darker, multi-layered and aggressive timbres always lies just a few steps away. Even the simplest wavetables will morph in milliseconds into edgy, distorted territory. The granular oscillator, meanwhile, absolutely exists to do hazy and splintered. The FM/AM section is another source of wild and wonderful textures, and that's even before you've fired up the Polar wavefolder and distortion effect.

Of course, I'm describing programming here. If you're more of a preset person you may think you've arrived in synth heaven when you start mining the in-built and online Minimal Audio reserves. Many presets have a colossal wow factor, and often a really contemporary 'produced' feel, as if they were sections lifted from productions you'd worked on for a week. Lots seem tailor-made for anything from EDM, trance, trap and jungle to cinematic soundtrack and more experimental electronic use, and will require only a single note to be played for sophisticated results to be heard. The mind-boggling timbral range that gushes out of Current has to be heard to be believed: in part it's down to that amazingly versatile synthesis architecture, with sequenced and arpeggiated elements, but probably at least as much the way the synth draws on Minimal's existing commercial sample libraries in the granular and sampler oscillators.

If there's one down side to this often larger-than-life, even bombastic character, it's that the provision of simpler vintage timbres is much less good. For example, I worked through all 160 locally accessible presets with the tag 'bass' and I'd say less than 10 sounded like they could have been generated by a Minimoog or other analogue synth and used to play a bass line in a conventional way. Many of the rest were banging, howling, digitally degraded mini masterpieces of their own that could spawn a composition, but might struggle to fit into an existing one. It was

Current Contract

When I first crossed paths with Current, at the moment of its release in October 2023, it was only available via an ongoing monthly or annual subscription. That seemed a particularly brave strategy with the Waves plug-in debacle having settled down only months before, and within only days the pricing/purchase model had expanded to include other options. Here's how it works now.

A perpetual licence for Current costs $199, and that gives you the synth and its factory/offline content, just like most competing 'disconnected' products. However, you also get a year of Minimal's 'All Access' plan, which is both the access to their online content and the substantial additional perk of all the synth's effect processors being available to download as individual plug-ins, so you can use them elsewhere in your DAW, independently of the synth.

The other option is a subscription, which actually is just as much a rent-to-own scheme. This is the 'All Access' plan pure and simple, costing $15/month or $120/year. It includes the same benefits as the perpetual licence (online content, plug-ins, etc), but also includes 'store credit' equal to what you spend, with annual subscribers getting a bonus $60 credit top-up at the end of the year. Store credit can be used to buy perpetual licences from Minimal Audio's online store — and that includes Current itself. The official line is that it would take you 14 months on the monthly plan to amass enough credit to buy a perpetual licence for the synth, which could be a rather nice, smooth way to acquire it. You could of course use store credit to buy Minimal's plug-ins or sound packs, though.

It's good that Minimal Audio offer perpetual licensing alongside subscription: people like to work in different ways. But the All Access plan is involved regardless of which you choose, and what happens to your investment if you let the plan lapse may not be immediately obvious.

For perpetual licence owners, you'll lose access to Minimal's online feed when your All Access runs out. You can still create new instances of Current, but you won't be able to load into them anything other than the embedded/local presets, wavetables and samples. Interestingly, though, Minimal confirmed that existing instances of the synth (in your DAW projects) that used online content will continue to load it. Similarly, instances of Minimal's individual plug-ins on DAW channels will also still instantiate and work just fine, but you won't be able to tweak them.

For users that only ever subscribed, this listen-but-don't-touch policy extends one stage further, to Current itself. Old instances of the synth will apparently still open in your DAW projects, and again will still load previously-accessed online or local material, but you won't be able to interact with them at all.

I wasn't able to test the robustness of any of these fall-backs, but if they work as promised it seems an eminently reasonable solution to a potentially thorny problem, and which should prevent users from finding themselves high and dry with silent synth tracks and dummy plug-ins, whether they've let their All Access sub lapse by design or by mistake.

Minimal Audio also tell me that future incremental updates to Current (to maintain compatibility with new operating systems and so on) will be free to anyone who's ever used it, via licence or subscription, whether their plans have lapsed or not. So all in all it's a fair and apparently well thought-out situation, that avoids the cliff-edge you're dumped over when unsubscribing from something like Adobe's Creative Suite, for example.

a similar situation with leads and pads, which often tend towards the huge and cinematic: chock full of movement and even embedded harmonic progressions (and yes, this aspect in 'leads' too!), and not exactly team players. It's good that a synth exudes character, but I wonder if Current could have wider appeal if it's eventually equipped with some more generic sound packs and conventional preset types over time. It's certainly more than capable of supporting them.

Currentcy

In the final reckoning, and by almost any standards, Current is an absolutely phenomenal synth. I grew to really love using it during the review period.

Drawbacks are few. It can be CPU intensive, especially with all synthesis sections and a slew of effects enabled. Nothing that the most up-to-date CPUs and a bit of judicious track freezing can't handle, but something to be aware of if you're planning to fire up 25 instances.

Then there's the polyphony situation: every instance is eight-note polyphonic at max. Not terrible, especially compared to broadly similar hardware synths like the original Waldorf Quantum, and a lot of the time you'd never notice, especially when triggering any of the factory one-key symphony presets. But it seems a surprisingly limitation in software, and becomes an issue most of all with long release times, where you might hear note stealing.

The user interface is, it has to be said, rather spartan: grey and purple (with a smattering of yellow) is the new black here. These things are very much in the eye of the beholder, though, and the flip side is that the upper-case bold-type labelling makes for great clarity, and it's impressive that almost all synthesis parameters can be visible on a single screen, with modulation paths shown clearly and intuitively. One thing I was less keen on is the documentation: just a handful of quite cursory pages on Minimal Audio's web pages, supplemented by video walkthroughs. They're good for learning the basics fast, but I'd take a PDF (as well) any day. Actually, a mouse-over tooltip mode is pretty good, doesn't get in the way like some do, and helps not to have to move away from your DAW at all.

As for the positives, I could go on and on. It's just so versatile. And I'm quite glad it is a true synth, and doesn't try to be a complicated scripted sampler like NI's Kontakt or UVI's Falcon. That keeps it feeling immediate and manageable.

As for the online content feed, yes, that does generally have a strong character and attitude. If it's not quite your thing you're not forced to use it, but if it is you might feel like you're set for life. There is a little bit of complication surrounding the All Access plan generally, that even hardcore perpetual licence lovers will end up dealing with, but Minimal's policies for post-subscription life are well considered and support the user, which is a refreshing change.

It's not the cheapest soft synth out there, but Current is, currently, amongst the very best. ■■■

$ See 'Current Contract' box.
W www.minimal.audio

Gauge ECM-87 Virtual Mic Locker Kit

Microphone Modelling System

Gauge's affordable ECM-87 has a virtual dimension.

PAUL WHITE

Gauge are a US-based microphone manufacturer whose products are hand-soldered in the USA using parts from China, Japan and Germany. Assembly is followed by an intensive QC stage, also in the USA, before shipping. We looked at their valve ECM-47 back in *SOS* December 2022, and here we're reviewing the solid-state, transformerless ECM-87. Both are large-diaphragm capacitor microphones loosely based on vintage German designs, but the ECM-87 model has a twist: it can be used with Gauge's Mic Clone plug-in to emulate a range of other classic microphones.

Starting with the mic itself, which is available on its own or as part of the Mic Locker Kit bundle, the ECM-87 features a very Neumann-esque outline and is built around a one-inch, cardioid-pattern capsule skinned with six-micron-thick membrane material. Its specifications are comparable to those of other similar microphones, and include a sensitivity of 12.5mV/Pa, a maximum SPL of 128dB at 1kHz and an equivalent noise level of 17dB (A-weighted). A standard 48V phantom power source is required for use.

Unscrewing the base ring allows the satin chrome body sleeve to be removed, showing two neat circuit boards populated with full-sized rather than surface-mount components. A switch on one of the circuit boards activates a low-cut filter. While this is less convenient than a switch mounted on the mic body, the low cut can usually be left engaged if the mic is being used for vocal recording or instruments that don't project a lot of low end. The data sheet that comes with the mic suggests that it can be used to record just about any instrument in addition to the human voice.

All the metal parts, other than the body sleeve, have a bright chrome finish. The mic comes with a soft storage pouch and a metal-framed, elasticated shockmount.

Let's Talk About Specs

While the quoted 20Hz-20kHz frequency response figure doesn't tell us anything particularly useful without any limits being specified, the response plot is more revealing. In essence, the mic is nominally flat up to around 2kHz, with no LF roll-off apparent before the graph stops at 20Hz. Above 2kHz the response rises to a first presence peak at 4.5kHz before dropping back between 6 and 7 kHz, then it climbs again to a second peak at around 12kHz. The lower peak extends to 4dB above nominal while the higher peak maxes out at around +6dB. This is somewhat different from the response of its European inspiration, which is much flatter in the presence region.

Moving onto the software component, the Gauge Mic Clone plug-in comes in AU, VST2/3 and AAX formats for macOS and Windows, and was developed by Gauge's Dr Chandler Bridges and his research team, in collaboration with Final Mix Software.

As far as I can tell, the Mic Clone plug-in works on the 'match EQ' principle, modifying the output of the ECM-87 to present the same frequency responses as a range of classic mics. This is not the first time a company have come up with a way of making one mic sound like another, but it is one of the most affordable examples. The plug-in is authorised using an iLok account, but there is a seven-day free trial. You can also buy the mic and software as a bundle.

The classic mics modelled are the Neumann M49, U87, U67, U47 and U47 FET,

Gauge ECM-87 Virtual Mic Locker Kit
$399

PROS
- A very capable large-diaphragm mic.
- The optional Mic Clone software is tailored to the ECM-87's response for emulating other mic models.
- Shockmount and storage pouch included.

CONS
- Low-cut filter switch is inside the mic body.

SUMMARY
While there are countless Chinese-built microphones in this price range, the Gauge ECM-87 gives a good account of itself and has the advantage that it can be used with the Mic Clone software to give it a range of classic voices.

The Mic Clone plug-in offers a range of classic microphone emulations.

AKG's C12 and C414, and the Sony C800G — a good cross-section of go-to studio mics. Essentially the plug-in applies an EQ curve that is the difference between the response of the ECM-87 and the target microphone. A fader allows the sound to be morphed gradually from unprocessed to processed, so you can also explore the inbetween sounds. The mics are fully tone matched at the fader's mid point: go further, and you get into 'nothing succeeds like excess' territory. This approach to tone matching has the limitation that it can only work correctly for on-axis sounds. No information is provided as to whether the plug-in replicates any of the target mic's saturation characteristics, which would be relevant in the case of valve microphones.

As the Mic Clone plug-in takes the ECM-87's frequency response as its reference point, it won't produce the correct effect with other microphones, though you may still get interesting and usable results. However, Gauge also offer a similar plug-in called Mic Locker, which is designed to to coax alternative tonalities out of any microphone.

ALTERNATIVES

The idea of pairing a microphone with software to allow it to emulate other models isn't new, but the competition tends to be rather more expensive than this Gauge example. Alternatives include the **Universal Audio SC-1**, **Antelope Audio**'s **Edge** range, and of course the **Slate VMS**.

Model Behaviour

Not having access to the mouthwatering selection of classic mics emulated by the ECM-87 Mic Clone package, I had to evaluate the sounds of the mics on a purely subjective level. Starting with the M49, this setting pulls back some of the presence of the raw ECM-87 sound so may be a good choice for those with naturally sibilant vocals, while the U87 model flattens out the response to some extent, making for a natural balance. The U67 is similar, with perhaps a hint more clarity and a solid sense of tonal weight, while refraining from sounding aggressive in the highs. This was my favourite setting for male voices that needed help in the lower mids, and it also works well on electric guitar. The U47 adds warmth, while the 47 FET keeps the weight but also captures transient detail well. The C12 is a real classic, and this emulation sounds transparent and lively without being too cloudy at the low end. I have a C12 VR (admittedly not quite the same thing as a C12), and this C12 emulation has something of the same low-midrange lightness that my mic exhibits. I'd describe the sound as open, with smooth highs, but not at all weighty. AKG C414s come in various flavours, but the model offered here has a voicing similar to that of the C12, with a very open high end.

Depending on the sound source, the differences between Mic Clone settings can be quite subtle, but I'd suggest that rather than obsess about replicating a particular mic's sound, you simply try all the options to see what sounds best with the voice or instrument you have recorded. It is often the finer details in the response curve of a microphone that either flatter a voice (or instrument) by emphasising what sounds good about it or, in some cases, what is less desirable, in which case it is time to move on to another mic model!

The great thing about ECM-87 Mic Clone is that it allows you to experiment with different microphones after the fact. The Gauge ECM-87 is a very capable mic in its own right, the Mic Clone software really extends its usefulness — and the whole package is very affordable. ■■■

$ $399 (discounted to $349 when going to press).
T Gauge +1 855 424 2843
E info@gauge-usa.com
W www.gauge-usa.com

HENDY AMPS
Michelangelo
Experience the raw sonic beauty and inspiring ease of use from Hendyamps' renowned all-tube EQ supercharged with modern digital capabilities.
Get instant tube vibe and effortless tone sculpting combined with precise band control, dynamic EQ with transient/body separation, sound stage enhancements, and deep customization of the tube circuitry - all with the Michelangelo sound.
TONE PROJECTS
TONEPROJECTS.COM

Remic Reshape

Instrument Microphones

Should you buy a specialised instrument mic, or a multi-purpose 'pencil' design? With the Reshape series, you can have both...

BOB THOMAS

Based in the Danish town of Silkeborg, Remic specialise in the design and manufacture of musical instrument microphones. Founded by artist, musician and engineer Thorkild Larsen in 1996, the company's research has yielded a highly regarded range of instrument-specific miniature capacitor microphones for grand piano, bowed strings, brass and woodwind. The Reshape RE7100 and RE7200 electret microphones form the company's first new product line since Larsen's departure in 2020, and they mark a change of strategy. These two microphones are considerably larger than their predecessors, and their instrument-specific mounts are sold as separate items, along with a 'pencil 'mic' adaptor that fits into a standard microphone clip.

Remic Reshape
$640

PROS
- Both microphones deliver impressive, great-sounding results.
- Instrument-specific mounts.
- Simple and effective 'pencil mic' adaptor allows them to be used as conventional standmounted mics.

CONS
- No acoustic guitar mount available.
- Violin mount doesn't suit every instrument.

SUMMARY

These new Reshape microphones from Remic offer the discerning musician and engineer distinctive and great-sounding stand- or instrument-mounted alternatives in both studio and live environments.

Black Is Black

Clad in Remic's classic all-black livery and sitting at the end of a 2m long black, cotton-clad cable, the two new microphones share the same form factor: an approximate overall length of 59mm (including the cable strain relief) and a diameter of approximately 8.5mm. The mics' 8g metal bodies take up the first 29mm or so of their total lengths with their (as best as I can tell) 5mm capsules positioned behind slotted grilles. The impedance converters of both microphones sit inside the cables' male XLR connectors and are powered by 6-48 V phantom power.

The RE7100 features a pressure-operated, omnidirectional capsule that picks up sound arriving from all angles evenly. The microphone's polar pattern isn't entirely symmetrical, as there is a +4dB bias towards the front, which won't cause any issues in its intended applications. The RE7200, by contrast, is a supercardioid mic and thus rejects sound arriving off-axis, albeit with a slight rear pickup lobe.

Being Specific

Both microphones have a stated frequency range of 20Hz-20kHz. The frequency response curve of the RE7100 (measured at 15cm from source) shows a fairly flat response up to 1kHz and then a gentle rise to a peak of +6dB at 11kHz, dropping back to 0dB at approximately 19kHz. The RE7200's frequency response covers the same range, and shows a flat (±1dB) response to 1kHz, followed by a gentle rise to +5.5dB at 7kHz, which then drops to -8dB at 19kHz or thereabouts.

The RE7100 can cope with 125dB SPL and the RE7200 can withstand 128dB without exceeding 1% total harmonic distortion (THD). The RE7100's A-weighted signal-to-noise ratio (SNR) of 68dB equates to a very respectable self-noise figure of 26dB and a dynamic range

of 99dB at less than 1% THD. Similarly, the RE7200's 66dB SNR equates to a self-noise of 28dB and a dynamic range of 100dB at less than 1% THD.

Instrument Mounts

The Reshape mounts, which fit both the RE7100 and the RE7200, follow the same form factor and compression mounting paradigm as Remic's current series of instrument microphones. The biggest difference this time round is that instead of a microphone and its housing being permanently integrated into an instrument-specific mount, either microphone can be fitted into any mount. This change makes a lot of practical sense in that, for example, a multi-instrumentalist, recording studio or hire company could maintain a stock of Reshape microphones and mix and match those with mounts as required.

As with their earlier instrument-specific equivalents, the Reshape mounts for violin, viola and cello are designed to be mounted under the ends of fingerboards, with the double bass mount's recommended mounting point being under the tailpiece. Naturally, there is no requirement to follow those placings and you may find that your cello sounds better with the microphone under its tailpiece; or that on double bass, mounting the mic under the fingerboard or even under the bridge itself is more to your taste. All Reshape mounts come in pairs, making the pricing a bit less painful.

The CE7000 cello and BA7000 bass mounts are pretty much identical in overall appearance and size to their predecessors and, as before, the major differences between them are their overall dimensions, the number of decorative cut-outs involved, and the size of their accompanying circular pads that can be fitted around the microphones' cables in order to hold them securely in position under either the fingerboard or the tailpiece.

The VI7000 and VA7000 mounts, for violin and viola respectively, have been significantly revised to carry the Reshape microphones. As before, the only difference between the two is that the viola mount is larger. The earlier type of mount had a set of laterally oriented grooves that sat in a wedge-shaped crest that sloped upwards from the back of the mount to the front. These grooves — and the pliability of the open-cell foam that made up the crest — allowed you to fit the mount quite easily under the fingerboard no matter the angle and distance between the fingerboard and the front of the instrument.

In contrast, the Reshape equivalents are made of a much denser and stiffer closed-cell foam, and the laterally grooved crest has been replaced by longitudinal grooves arranged in a relatively gentle horizontal arc. This change means that the mounts have to be compressed to fit under a violin fingerboard, making fitment more difficult than before — especially with a non-compressible metal cylinder occupying half of the mount's central area and leaving relatively little foam to compress in that area.

The BR7000 brass instrument mount again follows the same paradigm as its earlier equivalent. Made of natural rubber, its density, rigidity and circular Pac Man-style profile allow this mount to attach firmly to the bell of a saxophone or any brass family instrument by gripping it in its 'mouth'. A square-ish extension, positioned at a slight angle to the mouth, holds the microphone firmly in a tangential orientation that points the microphone into the throat of the instrument.

The completely new mount in the line-up is the PH7100 Pencil Holder, which, to my eye, resembles a long-necked wine bottle. Its major feature is a lengthways slot that allows you to drop the mic cable into the adaptor. You then pull the microphone backwards into the adaptor and hold it in place with the small, cable-mounted foam cylinder that you'd normally jam under the fingerboard of a violin or viola.

Mount Up

Mounting the microphones on my wife's late 19th Century French violin threw »

The PH7100 Pencil Holder lets you use a Reshape just as you would a standard small-diaphragm capacitor microphone.

The VI7000 mount fits some violins better than others.

» up issues that I didn't expect. Firstly, I couldn't fix either microphone securely in place without having to use quite a bit of pressure to compress the mount enough to get even half of its width to remain under the end of the fingerboard. Unfortunately, the compression pressure trapped between the fingerboard and front negatively affected the violin's character and tone. Also, with around 7mm of mount sticking out from underneath the fingerboard, once the RE7200's lateral vents were clear of the mount, the capsule ended up sitting too close to the bridge for my liking. I could position the RE7100 flush with the end of the mount, but that created a lateral instability that took a bit of cable repositioning to resolve.

Having said that, the violin mount turned out to be too low to fit securely under the fingerboard end of a more modern violin, although a cut-down viola mount would have worked perfectly. I think that Remic might want to do some more work on this particular mount. Although there isn't a viola in the household, there is an octave violin, which the viola mount fitted perfectly. The cello and double bass mounts were similarly simple installations and, as you'd expect, the pencil holder adaptor offered no challenges.

ALTERNATIVES

Other than the models from **DPA** and **Neumann** mentioned in the review, there's no other direct equivalents that I am aware of. However, although you'll find similar performance levels from both **MB Microphones**' standmounted **MBC 603** body with 5mm **KA100** omni or **KA500** hypercardioid capsules and **DPA**'s **d:vote Core 4099** range, I can't find other microphone systems offering the flexibility of the Remic Reshape series.

In Use

In terms of their overall sound, both the RE7100 and the RE7200 performed extremely well, delivering a detailed and dynamic sound with superb transient definition from every instrument that I tried them on. Being well used to my own DPA 4099's +2dB lift at 10-12 kHz, I thought I might find the Reshapes' stronger boosts a bit too much. In practice, both delivered an attractive clarity, with the RE7100 being particularly successful. Due to its stronger emphasis in the 3-5 kHz region, and relatively steep roll-off above 8kHz, the RE7200 couldn't really match the open and airy sound of the RE7100, although it did provide what I'd describe as a less flattering representation of the source. Although I'd have no qualms about using the RE7100 on stage, the RE7200 would probably be my first choice in that situation due to its steeper high-frequency roll-off and ability to reject ambient sounds. One thing that particularly impressed me about these two microphones was that I never felt that I would have to use EQ to get my instruments to sound like themselves — even on cello and double bass, where both microphones delivered natural-sounding results with depth and clarity from both instruments.

I did find an issue on the violin with the RE7200 where, when positioned too close to the bridge as I described earlier, it appeared to be unable to handle the very high SPLs that are generated in that area, distorting audibly across all strings at normal playing levels. However, moving it to positions either above the bridge or over an F-hole removed the issue. The RE7100 didn't exhibit distortion at normal playing levels, probably because it could sit further back flush with the surface of the mount.

Sadly, my trumpet-playing days lie far behind me so I couldn't test either microphone on a brass instrument. However, in the absence of a specific guitar mount, I tried using the BR3000 brass mount to hold the RE7100 inside the soundhole of an acoustic guitar, where it worked very well and sounded great, despite its non-ideal positioning.

Stand-mounted in the pencil holder, both mics turned in uniformly excellent performances as conventional instrument microphones across a wide range of stringed and percussion instruments, showcasing their versatility and general-purpose possibilities.

Summing Up

With their instrument-specific mounts and pencil holder adaptors, the Reshape RE7100 and RE7200 are being promoted by Remic not only as premium instrument microphones for stage and studio, but also as high-quality, general-purpose microphones. In both market sectors the Remic microphones will be competing directly with very similar products from more established companies. For example, as a standmounted omnidirectional studio microphone, the RE7100 in its pencil adaptor is the only product that I know of that can compete directly with the DPA 4090 in terms of its capsule diameter, price and performance. Similarly, in the instrument mounted microphone sector, the RE7200 will find itself facing direct competition from Neumann's MCM System.

Overall, I was very impressed by the sound and performance of Remic's RE7100 and RE7200 and, in my opinion, each offers the discerning musician and engineer distinctive and great-sounding alternatives to their competitors at a price point that reflects their quality.

$ RE7100 & RE7200 $640 each, instrument mounts $75 each.
E info@remic.dk
W www.remic.dk

fabfilter
software instruments

Engineering Emmy® winner

Pro·Q 3

Unrivalled sound and workflow

An equalizer is probably the tool you use most while mixing and mastering, so you need the best of the best! With FabFilter Pro-Q 3, you get the highest possible sound quality and a gorgeous, innovative interface with unrivalled ease of use.

Distributed by Music Marketing Inc.

To find a dealer visit **www.musicmarketing.ca**

Amphion One25A

Active Monitors

We put Amphion's first ever three-way design to the test.

Amphion One25A
$14,900

PROS
- Spectacular bass.
- Utterly revealing of mix detail.
- Perfect tonal balance with minimal coloration or distortion.
- Hugely enjoyable.

CONS
- Expensive, big and very heavy.

SUMMARY
The Amphion One25A eschews DSP and relies on more traditional electro-acoustic design and engineering to create a truly outstanding high-end active monitor. It ranks among the very best.

PHIL WARD

Finnish speaker and amplifier company Amphion have carved out a niche in the professional monitoring market with their much admired range of two-way nearfield speakers, such as the Two15 I reviewed back in 2017. Their reputation for thoughtful and innovative electro-acoustic engineering is well deserved, and their approach translates into highly effective monitoring. But it has always felt as though there were a couple of elements missing from the Amphion range: active drive, and a three-way speaker. Of course, Amphion have had their own range of amplifiers for a while, and subwoofers too, and these arguably fill the gaps. Now, though, they've brought everything together into the subject of this review: the three-way, active One25A.

Before you get too excited, a couple of health warnings. Firstly the One25A is not an inexpensive monitor (I'd deploy the term 'aspirational') and secondly, it is very much at the large end of the nearfield monitoring spectrum — it's a midfield, really. At 41kg, it is also outrageously heavy. So heavy, in fact, that instead of testing them in my normal garden studio and large acoustic measuring space, I had to take a different approach. My local recording and rehearsal complex, Brighton Electric, very generously offered their Studio 2 control room for listening.

First Look

The One25A's visual appearance is unmistakably Amphion, and none the worse for that. I've always admired the simplicity of the Amphion aesthetic, with its combination of dark, matte cabinet surfaces and aluminium driver diaphragms set off by the whiter-than-white tweeter waveguide. It has the look of a high-precision, professional tool, and I rather like that.

Its dimensions are 316 x 510 x 487mm (HWD). Compact, it's not. And if you want to know why it's so heavy, as well as its cabinet being constructed from 25mm thick, heavily braced MDF, the bass driver alone weighs 10kg. The chassis of the bass driver is even incorporated into the cabinet bracing. Furthermore, the One25A also incorporates numerous constructional measures designed to ensure that the midrange driver and tweeter are mechanically and acoustically isolated from the bass driver. The narrow perforated grille on the front of the enclosure, for example, terminates a foam-filled air-gap slot that runs diagonally through the cabinet from the front through to the rear side. The slot and its internal damping ensure separation between the driver elements of the One25A. Furthermore, the diagonal geometry of the slot results in the separate bass and midrange enclosures being asymmetric, which helps discourage internal standing waves. And when I asked Amphion about the weight of the One25A, the response was that it wasn't really something they considered. When you set out to make a no-compromise active monitor, it weighs what it weighs.

Bolted to the rear panel of the cabinet is a filter, EQ and three-way amplification module housed in a large folded-steel enclosure. The amplification is rated at 205W each for the mid driver and tweeter, and a generous 700W for the bass driver. So, even though Class-D technology is known for its light weight, amps supplying a total 1.11kW were never going to be featherweight. The crossover filters are all fourth-order (24dB/octave) types, and rather than employing active op-amp chips, are implemented using passive networks buffered on their inputs and outputs. The whole electronics module is removable to enable the monitors to be soffit-mounted, and Amphion are additionally planning a rackmount version of the module. On its underside are a mains power input and switch, a balanced XLR input, and a stepped knob that offers a ±8dB range of LF equalisation profiles to provide some compensation of low-frequency level depending on the monitor's installation with regard to room boundaries. The electronics module offers no other connection or configuration facilities.

The Low Down

Like the bass/mid drivers in Amphion's passive two-way monitors, the One25A bass drivers come from Norwegian specialists SEAS. The bass driver is a nominally 25cm (10-inch) unit, designed specifically for low-frequency duties alone. That's clear from the extremely generous roll-surround fitted to the driver and the fact that its motor system (magnet, pole-piece, top plate and voice coil) provide ±14mm of linear diaphragm excursion — around twice that of smaller bass/mid drivers. But the driver's motor system is not only impressive in terms »

» of diaphragm excursion: its voice-coil is also unusually large at 56mm in diameter (getting on for twice the more usual 30mm). The driver is clearly designed to generate low bass at high volume levels with minimal compression.

And if that wasn't enough, the motor system also incorporates a copper cap on its pole-piece, which functions to reduce the voice-coil inductance and the degree to which inductance changes with voice-coil movement. I'll unpack that a little more. Voice-coil inductance results in a resistance to the flow of electrical current that increases with frequency, and in many speaker drivers, the inductance changes depending on the position of the voice coil. And as voice-coil inductance influences a speaker's frequency response, having it change in response to the input signal (because it's the input signal that makes the voice coil move) means that the input signal can modulate the response — which, you probably don't need me to tell you, isn't a good thing. Making sure inductance modulation is minimised is particularly important on a driver designed for high levels of diaphragm excursion, so the copper cap of the One25A bass driver is a valuable refinement.

But why does the One25A need a bass driver that offers very high diaphragm excursion potential? There's two related reasons. The first is that the One25A's specified low-frequency bandwidth is -3dB at 22Hz. 22Hz is subwoofer territory, and without the help of reflex loading (the One25A is a closed-box monitor) the bass driver is very much on its own. The context here is that the driver excursion required to generate a constant sound pressure level increases rapidly as frequency falls. For example, all other things being equal, 90dB (at 1m) at 100Hz requires around ±1mm of excursion from a nominally 25cm diameter diaphragm, but the same 90dB at 20Hz requires around ±4mm.

The second reason for the generous excursion capability is that its 22Hz low-frequency cutoff isn't achieved simply by mounting in the cabinet; it requires equalisation. By my rough calculations, without low-frequency EQ, the One25A would display a -3dB cutoff at around 50Hz, with a 12dB/octave fall in output below that. So, to reach a -3dB point at 22Hz, nearly an octave lower, the One25A needs around 10dB of gain below 50Hz, and that will put very significant demands on both diaphragm excursion and amplifier power. This explains why the One15A needs an LF amplifier rated at 700W: it provides headroom for the 10dB of LF EQ (and the further +8dB available from the user EQ). Finally, I rather glossed over the closed-box loading, but of course this is hugely significant in terms of its low-frequency behaviour in the time domain. Group delay (low-frequency latency) will be low (probably around 5ms), and low-frequency transient signals will stop when they are supposed to, rather than being effectively extended by the reflex port resonance. There's also no reflex port to introduce compression, distortion or noise as volume levels rise or to impart uncertainty to low-frequency pitch.

State Of Flux

The midrange driver is also sourced from SEAS in Norway, and is closely related to the drivers employed in Amphion's well known two-way passive monitors. It's a nominally 130mm-diameter driver with an aluminium diaphragm and, like the One25A bass driver, has a sophisticated motor system designed to minimise the distortions inherent to moving-coil drivers. Again, a copper element is employed to counter a modulation effect but, in this case, it's a copper ring around the pole piece rather than a copper cap, and its job is to suppress a phenomenon known as flux modulation (conducting rings in driver motor systems are sometimes known as 'shorting' or Faraday rings). Flux modulation describes a phenomenon in which the input signal creates its own magnetic field that modulates the fixed field of the driver magnet. As with inductance modulation, flux modulation will be imprinted on the driver output as distortion, so measures taken to stop it happening, such as the midrange driver's copper ring, are of significant benefit.

One respect in which the One25A midrange driver is slightly atypical is that it employs a large roll surround, of dimensions that would normally suggest bass duties. And that's because the One25A employs an unusually low (100Hz) bass/midrange crossover frequency. This is getting on for two octaves below that of a typical three-way monitor crossover. I referred earlier to the One25A's subwoofer credentials, and with its bass driver low-pass filter set at only 100Hz, 'subwoofer' is the appropriate description! This also means that the midrange driver's role is more towards bass/midrange duties because, despite its relatively steep high-pass active filter slope of 24dB/octave, its output will only be around 12dB down at 75Hz.

The reason Amphion employ such a low bass/mid crossover frequency has to do with managing system directivity, in effect configuring the three drivers of the system to work as a point source. I'll describe how this works in terms of the midrange and tweeter a bit further down, but in terms of the bass and midrange drivers, the low crossover frequency ensures that in the region where the output of the drivers overlap significantly (let's say an octave either side of 100Hz), the wavelength remains much longer than the physical distance between the drivers. In this particular case, the wavelength at 200Hz is 1.7m and the drivers are around 0.3m apart. Those numbers taken together mean that off-axis path length differences from the drivers to the listener (or a measuring mic) don't diverge by any significant portion of a wavelength, so the off-axis frequency response doesn't suffer from interference dips. If the drivers were further apart, or the crossover was significantly higher, path length differences would result in destructive interference between the drivers and consequent dips in the near off-axis frequency response.

"It's as if the One25A reveals the story of how sounds have been treated by the recording and mix process."

Prime Directive

Moving up the band to the midrange-to-tweeter crossover, the same kind of principles apply. The crossover frequency is 2kHz (wavelength 17cm) and the drivers are around 12cm apart. So although the equation isn't quite as clear-cut, the driver outputs probably remain reasonably in phase to around 30 degrees off-axis vertically. But there's another directivity factor that comes into play around the mid/high crossover, and that's the naturally narrowing dispersion of the mid driver towards the upper end of its band. This is primarily a function of the

mid driver's diaphragm diameter. Drivers naturally begin to become noticeably directional above the frequency at which their diaphragm dimensions are comparable to the radiated wavelength, and for the One25A midrange driver, that will be at around 1.5kHz. So, ideally, the midrange driver should hand over to the tweeter at around that frequency or a little higher, and that's the case with the One25A — its mid-to-tweeter crossover is at 2kHz. However, operating down to 2kHz would potentially be a power handling and distortion challenge for the relatively small (25mm) titanium-dome tweeter of the One25A, and that's where Amphion's signature UDD (Uniformly Directive Diffusion) waveguide comes into play.

The waveguide offers two really significant benefits. First, it provides an element of acoustic impedance matching for the tweeter that significantly increases its sensitivity, especially at the lower end of its operating band, and in doing so it neatly solves the 2kHz power handling and distortion challenge. I'd estimate that the waveguide results in an extra 6dB at least of tweeter sensitivity in exactly the frequency band where it's needed, and that's vital. The second benefit of the waveguide is that its diameter predominantly defines the directivity of the tweeter at the lower end of its operating band. So it's no coincidence that the tweeter waveguide and the midrange driver are of similar diameter. It means their directivity in the band where one hands over to the other is similar. When I talk of Amphion's "thoughtful and innovative electro-acoustic engineering", this is a perfect example.

Before I move on to my listening experience, there's one last element of the monitor to describe. Or rather, not describe, because in a world full of monitors defined by their DSP, the One25A remains free of digital intervention (at least in its signal path — its overload protection circuits are DSP-based). It is in many ways an 'old school' speaker, where performance is defined by the drivers and the skill with which they are integrated. Amphion's founder Anssi Hyvönen says he is not in principle against DSP in monitors, but he does believe that it ought to be subordinate to the electro-acoustics. He argues that the best performance is most likely to come from ensuring the electro-acoustics are optimised and that DSP is employed only for functions that can't be done otherwise. I guess the proof of that comes from listening...

ALTERNATIVES

If you're in the fortunate position where the One25A is a realistic aspiration, then you probably also ought to hear monitors such as the **Kii Three**, **Dutch & Dutch 8C**, **PSI A25M**, **PMC8-2**, **Genelec 8361**, **ATC SCM45A**, **ADAM S5V** and **Barefoot Sound MM26**.

Studio Time

Before listening to the One25As at Brighton Electric, I spent some time familiarising myself with the space by listening to the monitors already installed. Coincidentally, the monitors in Studio 2 are of a design and size not hugely dissimilar to the One25A. They would probably even be seen as a competitor. They sounded great — both in terms of enjoyment and their likely use as an analytical mix tool. I spent an hour or so with them, playing a whole bunch of well-known pieces, then took them down and installed the One25As in their place.

The first thing that impressed, perhaps unsurprisingly, was the bass. I began playing one of my regular reference tracks, 'Sycamore' from John Metcalf's *Appearance Of Colour*, and almost immediately forgot that I was supposed to be listening critically and became drawn into simply enjoying, and appreciating anew, Ali Friend's wonderfully sinuous and inventive double bass lines. One25A bass is hugely extended in terms of bandwidth and massively powerful, but simultaneously very fast and dynamic, without the slightest hint of pitch uncertainty or resonant overhang. And it doesn't really seem to care about volume level; sensible or really quite loud, the One25A remains consistent and able to resolve and make audible the smallest low-frequency detail. The One25A's low end provides an utterly secure foundation for everything above, and I can't really imagine a scenario where I'd want any more bass extension or quality from a nearfield or midfield monitor.

It's a similar story further up the frequency band. Mid-band voices and instruments are handled with an unforced, neutral tonality that somehow makes irrelevant any thoughts of "too bright" or "too dull" in monitoring balance terms. Simply recorded voices just materialise in space, fully formed and focused, sounding convincing such that any mix artefacts of compression or reverb are explicitly revealed — they sound almost separate to the voice. It's as if the One25A reveals the story of how sounds have been treated by the recording and mix process. You don't just hear the final result, you hear how it came to be.

The tweeter just continues the work of the midrange driver, delivering an integrated whole with masses of easy high-frequency detail and clarity. One of my regular listening techniques is to evaluate the balance in naturally recorded voices between vowels and consonants. Does it sound natural? Is it believable? Is the balance obviously modified by compression or EQ? If a monitor's mid and high bands are well balanced, and well integrated in terms of timing and directivity, the vowel/consonant balance should sound convincing if the recording is natural, or symptomatic if it's been messed with. Presenting this balance accurately is, to my mind, both a vital ability and a good indicator of useful performance in a monitor, and the One25A possesses the ability to a level that is right up with the best.

Conclusion

I spent rather longer listening to the One25As than I really needed to establish its credentials — in truth, it was obviously something special from the first few bars. But listening was such a pleasure, and I had the loan of Brighton Electric's control room right through to the end of the day, so I even went back in the evening to listen some more. The One25A does that: draws you in and doesn't let you go. Of course, just because a monitor is enjoyable, that doesn't always make it an effective mix tool — sometimes flawed speakers are the most fun — but that isn't the case with the One25A. It's an incredibly accurate, revealing and capable mix tool over a massively wide bandwidth and at pretty much any volume level. It really does perform up to, and perhaps beyond, the level you'd hope for at the price.

Thanks to Brighton Electric for the loan of their Studio 2 control room for this review.

brightonelectric.co.uk

$ $14,900 per pair.
W www.amphion.fi

SPL Channel One Mk3

Mono Channel Strip

With a Transient Designer, a de-esser and an unusually versatile input section, there's more to SPL's recording channel than most.

MATT HOUGHTON

Sound Performance Lab (SPL) have been making high-quality audio gear since 1983. They first made a name for themselves with their Vitalizer, but it's probably their next innovation, the Transient Designer — the first dynamics processor that didn't rely for detection on the input signal crossing a level threshold — for which they're now best known. Now, SPL's pro audio range includes everything from preamps, channel strips, audio interfaces and mixing desks to mastering gear, monitor controllers and high-quality headphone amps, and with the last of those they also cater for the hi-fi market.

For review here is the third iteration of their Channel One. Like its predecessors, this analogue recording channel comes in a vented 2U 19-inch rackmount chassis, but while it borrows plenty from the Mk2 version, this is a significant redesign that goes much deeper than the darker and, to my eye anyway, more impressive and easier to read front panel. The most notable changes to the feature set include a revamped preamp section, and this is now joined by a dedicated valve-based saturation processor. The MkII's headphone monitoring facilities have been dropped too and although this was a high-quality feature, it's one that I suspect many users will have found superfluous. More than compensating for that is the inclusion of a Transient Designer, which should increase this device's appeal and versatility significantly. The de-esser, EQ and compressor sections seem largely unchanged, but the metering has been rethought: the previous version had LED meters for gain reduction and output level, whereas we now have a large moving-coil meter that can be switched to show gain reduction, input level or output level. A switch sets this meter's 0VU position to correspond to output levels of +6, +12 or +18 dBu (that's the only means of user calibration). What's less obvious from the pictures is that SPL have opted for beefier power rails (±18V), and that the build quality on the inside is impeccable, with traditional through-hole components used throughout.

Ins, Outs & Amplification

At the start of every channel strip comes the preamp, and this one is more versatile than most. There are actually two versions of this device: the regular one is electronically balanced, while the presence of Lundahl input and output transformers differentiate the Channel One Mk3 Premium. The mic amp is a discrete solid-date design, but it's joined by a separate valve saturation processor. So, between the main gain control knob (9-68 dB for the mic inputs, and continuously variable) and the Saturation knob (turning this first switches in this circuit, then over 30 detented positions take you up to 100%), you can already access a range of sonic characters.

There's particularly generous flexibility when it comes to the inputs. On the back are two separate mic inputs (unusual for a mono device), and a dedicated line input, all on XLRs, while on the front

there's a high-impedance TS instrument input, which takes precedence over the line input when a jack is inserted. A toggle switch selects the source (Mic A, Mic B, Line/Inst), while separate switches for each mic input engage +48V phantom power. Three more switches operate a 20dB pad, a polarity inverter and a fixed 80Hz 6dB/oct high-pass filter. Though not indicated (that would have made the panel crowded) the gain range for line signals is -20 to +16 dB, and for instruments -6 to +30 dB. And if you have the input transformer option that adds a chunky 14dB to the values on the scale.

With the dual mic inputs and the saturation effect now being independent of the preamp gain, it's easy to compare the sound of two different mics (even if you must set the gain for each when you switch). I imagine the input setup could also appeal to the songwriter who regularly records two or three sources one at a time: you could have your go-to vocal and acoustic guitar mics plugged into the mic inputs, an amp modeller plugged into the line input, and patch a bass in the front whenever required: flip a switch, set the gain and you're ready to roll. No repatching required!

On the back, a preamp direct output is joined by two main outputs — the latter are identical, running in parallel — and all are XLRs. So you could, for instance, capture a clean signal from the Channel One when recording, and then return a line-level signal to the unit for mixdown processing. Or, since the preamp output is active at all times, you could record a clean signal as a backup and have the confidence to try more assertive processing. Or perhaps you want to capture a clean signal but use a processed one for a live-streamed broadcast. Or maybe you'd like both a clean and processed version for parallel processing... there's lots of potential. Also on the back, are a ground-lift button, a power switch, a voltage selector and an IEC power inlet. (Sadly there's no global power on/off on the front; it's now rare that I want everything in my rack on at once.)

One For All

As I said above, the Channel One Mk3 has plenty of processing options and each section on the strip, other than the preamp, has its own engage/bypass button. The valve saturation circuit is based around a Sovtek 12AX7LPS valve with a 250V anode supply and this is, by default, the next stage after the preamp, hence its position on the front panel. But a button (blue when engaged) beneath the VU meter can move it post-EQ (and pre the output stage). The valve circuit features automatic level compensation, and this proved so useful — the levels only start to creep up at extreme drive settings, and only by 6dB. The saturation sound can be beautiful, as you'd expect with a real valve, with subtle thickening distortion at the lower end of its range, and a pleasing 'flair' and 'crunch' when driven hard.

Next comes the de-esser. This has low (centre frequency 6.4kHz, bandwidth 4.4kHz) and high (11.2kHz and a bandwidth of 5.5kHz) buttons that illuminate yellow when engaged, and both can be active simultaneously. An S-Reduction knob sets the amount of de-essing from -0.5 to -12 dB, and ess detection (rather than the de-essing activity itself) is indicated by a single LED in the metering section.

SPL's famous Transient Designer section follows this on the faceplate and in the default signal path, and the controls comprise just an on/off button and two pots, each with centre detents for the neutral position. Attack can boost/cut »

SPL Channel One Mk3

From $2199

PROS

- Lovely, detailed preamp.
- High-voltage valve saturation circuit.
- How many channel strips include both a de-esser and a Transient Designer?
- Versatile I/O and routing options.

CONS

- Can't adjust compressor time constants.

SUMMARY

The Channel One has evolved, and the result is one of the most versatile channel strips around — of course, it also sounds great, and has tonal character on tap!

» transients by ±15dB, while the Sustain range is ±24dB. As with all SPL Transient Designers I've used over the years, this works very well, and I've loved having this sort of control on an all-purpose recording channel: it allows you to manipulate the character of anything with a percussive element (be it a drum, or the plucking or hammering of a string) in a way threshold-based processors cannot.

From here the signal enters the compressor, a low-noise and low-distortion dual-VCA circuit based around THAT 2181B ICs. This has only two control knobs. One, labelled Compression, is a threshold control that can be set from 0 to -20 dB, while Make-Up Gain can be set from 0-20 dB. You can't chance the time constants, so it wouldn't be my first choice for drums and percussion and I wouldn't recommend going overboard with this while tracking, but used sensibly on vocals and dialogue it sounded as smooth and unobtrusive as you might hope for in a recording channel.

The three-band EQ is the last processor in the default chain and on the panel, but it can be switched to come before the Transient Designer. The broad LMF bell band can be set anywhere from 30 to 700 Hz, while the MHF band, another bell, spans 680Hz to 15kHz. Both offer ±12dB of gain through centre-detented pots, while the frequency selectors are detented throughout to aid recall. The Air band is described as a coil-capacitor bell. This has a fixed centre frequency (19kHz) and offers ±10dB of gain, and as with all so-called Air bands, the extremes of this EQ curve reach well down into more easily audible parts of the spectrum. It enables you to subtly lift (or reduce) the sense of air or breathiness without things becoming too harsh (depending on the source, of course!). I found that it can be helpful in taming the brightness of some cheaper capacitor mics too.

More than most channel strips: as well as the preamp, compressor and EQ, the Channel One Mk3 features a de-esser and a Transient Designer.

Unusually, the Channel One Mk3 has two separate mic inputs for the same preamp, as well as a preamp direct out and two paralleled main outputs.

After the EQ there's an output level control, a pot and that runs from +6 to -20 dB, and a mute button, which does what you'd expect but also deactivates the meter. Finally, there's a single overload LED in the centre section — this lights up when an overload is detected at any point in the signal chain, not just the preamp or the output, which is a nice, thoughtful touch.

One Love?

A lot of brands now jostle for attention in the channel strip market. Curiously, I don't see so many opinions online about SPL as I do of many competitors — and really we should, because everything of theirs I've used has been good, classy-sounding gear, and the Channel One Mk3 not only continues that tradition but I'd say it's an improvement on what's gone before. Not only does it sound great to my ears, but there are real innovations here that give it capability not found (at least to my knowledge) in any other single device else. The ability to compare mics so easily makes it an outstanding candidate for anyone wanting a single, do-it-all channel strip, while having the preamp direct output available at the same time as two processed ones, and the ability to switch any stage in/out of the signal path and move some around, makes it yet more versatile.

So I'd use the Channel One Mk3 as a main vocal strip without hesitation. The clarity of the preamp and the controllable saturation really lend themselves to that, and the de-esser is a real plus — not many channel strips have this, and particularly when boosting higher frequencies with EQ, as is fashionable, they can be really helpful. The only real 'weakness' (if, indeed, it can be called that) is the compressor, as there will be some who crave more control. But it sounds good, plenty of people will enjoy the speed and simplicity of this 'two-knob' approach, and because it comes at the end of the chain, you could easily use a standalone one (or plug-ins).

This is by no means just a vocal channel, though, and not least because of that Transient Designer, which is a great bonus. Yes, there are plenty of transient shaping plug-ins now, including SPL's own, but while some offer more control I've yet to hear one that sounds as 'forgiving' as the analogue hardware versions, especially when a significant transient boost is called for. Combining the Transient Designer, tube saturation and compression can do wonderfully explosive things to drum sounds, whether kit pieces or, say, a room mic, although you do need to remember that this is a mono device and there isn't a way to stereo-link two units. What's more, you can control the result of such carnage using the EQ section and, to an extent, the de-esser too. Alternatively, you can deploy the Transient Designer much more conservatively. For example, I had great success using it to balance the pick and sustain on a fingerpicked acoustic guitar, before compression.

In short, there's not much here to dislike, and plenty to like. The preamp sounds clean and detailed, and it offers plenty of gain. There's real valve 'colour' on tap, and there are more processing and output options than on pretty much any channel strip of comparable quality or price. So if you're looking for a single 'do it all' channel strip, this one deserves serious consideration.

$ Channel One Mk3 $2199.
Channel One Mk3 Premium $2487.
W https://spl.audio

amphion®

Amphion One25A

Active full range studio monitor

Advanced acoustic design, honesty, and meticulous craftsmanship are core characteristics of all Amphion products. The newest addition to the family – One25A – is a culmination of uncompromised design choices to create an active monitor which meets our standards for sound. Sealed dual cabinet, refined signal path, DSP-free acoustic purity, and isolated electronics create results which need to be heard to be believed.

25 years of sound

MADE IN FINLAND

amphion.fi

IK Multimedia ARC Studio

Speaker Correction System

IK's monitor correction tech is now available in a standalone hardware box.

SAM INGLIS

IK Multimedia's ARC was one of the first affordable speaker correction products. The principle is simple: sine sweeps are played back through your monitors, and into a measurement mic placed at or near the listening position. An EQ curve can then be calculated to compensate for deficiencies in the response of the loudspeakers and, more importantly, the room.

In the first few iterations of ARC, the corrective EQ curve was applied in a software plug-in. This approach has many advantages: it's cheap to implement, easy to change on the fly, and there's no hard limit on the complexity of the curve that can be applied. It also has some obvious down sides, such as speaker correction being available only in your DAW and not to other programs, and the potential risk for mixes to be bounced through the plug-in.

But what's the alternative? Well, the correction could be done in a standalone, systemwide app, as is possible with Sonarworks' SoundID Reference, for example. Alternatively, it could be implemented in a dedicated piece of hardware that sits between your interface and your speakers; or it could be integrated into the speakers themselves.

IK branched out onto the last of these paths late in 2022 with their iLoud Precision MTM speakers, which have ARC built in. IK's existing measurement mic and software tools are used to calculate a correction curve, but this can be uploaded into the speakers' own DSP to fix the sound at source. They've now followed this up with a standalone hardware processor called the ARC Studio. Simultaneously, the ARC software itself has been updated to version 4.

ARC Story

Apart from the MTM implementation, ARC is now available in three progressively more costly variants. You can still buy the software alone, for use with a third-party measurement mic. You can buy the ARC software with IK's own MEMS measurement mic, as before. Or you can opt for the full package with MEMS mic and ARC Studio hardware, which was supplied for review. The ARC software is compatible with macOS and Windows, and is authorised using a serial number.

The MEMS mic looks much like any other measurement mic, except that it is pointed forwards in use rather than upwards. It has a standard XLR connector and needs to be used with a conventional mic preamp, which is not supplied. The ARC Studio box, meanwhile, has the same rectangular form factor as a typical small desktop USB audio interface; and, indeed, it has a USB Type-C port on it. I was surprised and a little disappointed to find, however, that it can't be bus powered: you'll need to use it with the supplied

IK Multimedia
ARC Studio
$300

PROS
- Slick, easy-to-follow analysis procedure.
- Offers both the flexibility of software and the convenience of hardware.
- Much more affordable than other hardware options.

CONS
- ARC Studio unit can't be bus-powered and has no digital I/O.

SUMMARY
The ARC Studio package combines a simple but effective hardware unit with powerful, intuitive software tools. If your monitors don't already have room analysis and correction features, it has the potential to make a big difference.

wall-wart PSU at all times. The rear panel also sports analogue input and output pairs on XLRs, and although the internal processing is digital, there's no digital audio I/O. Nor is there any provision for bass management, speaker switching or monitor control. The front panel of the ARC Studio is simplicity itself, with LEDs indicating power and signal present/clipping, and a single button to toggle correction on and off.

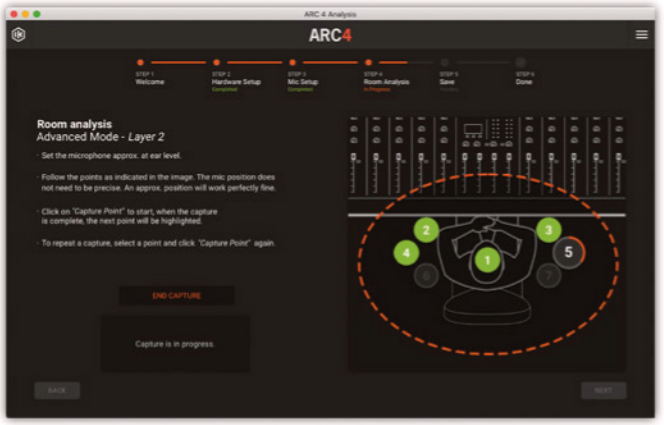
Like many manufacturers with a large software product portfolio, IK Multimedia have developed their own 'hub' app for downloading, installing and authorising products. I had some trouble persuading this IK Product Manager program to download the ARC software, but once I'd managed to get hold of the installer, everything was plain sailing. It actually installs two separate programs: ARC 4 Analysis, which carries out the room and speaker measurements and generates appropriate correction curves, and ARC 4 itself, which controls the ARC Studio box. The ARC plug-in is also installed, and could be useful for correcting secondary monitors even when you have the ARC Studio on your main pair.

Reading The Room

The ARC 4 Analysis app holds your hand pretty tightly through the process of measuring your room. IK's publicity material says that version 4 features an "all-new algorithm", and in order to get the best from this, it's now recommended that you measure at 21 separate points spread across three different height layers around the listening position. This isn't as arduous as it sounds, though, because absolute precision in mic placement isn't crucial, and there is the option to use fewer points if you're in a hurry. Once the process is finished, you can name and save the resulting curve ready to be loaded into ARC 4 and the ARC Studio.

The standalone functionality of the ARC Studio unit itself is as basic as its I/O. In essence, it can store one ARC 4 setting, which is retained even when your computer is switched off or the USB cable disconnected. In normal use, though, you may want to retain the USB link and keep the ARC 4 software running in the background, as it's needed to access most ARC Studio features.

Once you open up ARC 4, the first thing to do is to load in the profile that ARC 4 Analysis has created. This can be edited and adjusted in various ways, some more ill-advised than others. ARC 4 Analysis measures the left and right speakers separately and creates separate curves, but these can be combined into a single averaged correction curve, which is probably desirable unless you're working in a very unbalanced room. One very useful facility is that you can use 'window blinds' to exclude either end of the frequency spectrum »



The ARC Analysis software guides you through the measurement process in a very friendly and intuitive way.

DPA MICROPHONES

DDK4000
Drum Microphone Kit

DPA's selection of mics for the drum kit provides the most natural and precise sound possible, giving you complete control.

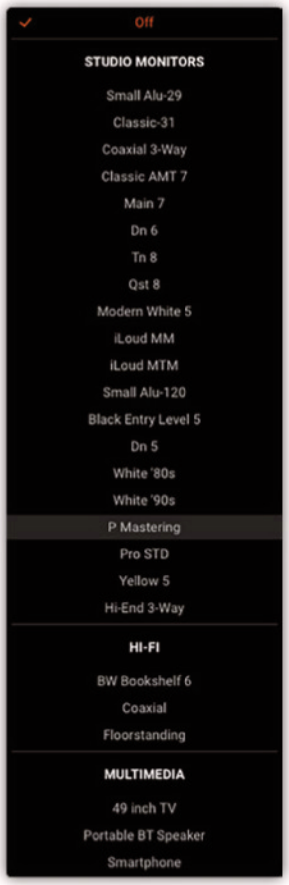
dpamicrophones.com

» from the correction. This will, for example, stop ARC 4 attempting to apply huge bass boosts to small monitors that aren't capable of putting anything useful out at 40Hz. It's now possible to switch between natural and linear-phase equalisation. A nice touch is that you can choose from a library of images of different monitor types to remind you that a particular profile is associated with a particular set of speakers.

Different people have different ideas about what constitutes the ideal monitoring balance, and whatever your own preferences, ARC 4 probably has you covered in its Target pop-up. As well as the obvious Flat setting, this allows you to impose bright or warm tilt EQs, a Dolby Atmos Target curve and more. You can also store and load your own custom curves here.

A second pop-up provides access to another ARC feature: speaker emulation. The idea is that, having painstakingly corrected the flaws of your own speakers and room, you can then introduce those of another system, such as the ubiquitous Yamaha NS10 or a consumer device like a TV or smartphone. It would be wishful thinking to expect too much from this, and it'll never make a cheap pair of monitors sound like a £100k mastering rig, but it can certainly be useful for checking mix translation.

Changes made in ARC 4 are heard immediately, but if you want to update the single setting that's stored in the ARC Studio for standalone use, you'll need to hit the Store button. This initialises a sync process that takes a few seconds.



ARC 4 offers numerous emulations of monitors and consumer playback systems.

Battle Of The Acronyms

My current main monitors, which I like very much, are Genelec 8330A actives paired with the matching 7350A subwoofer. These belong to Genelec's Smart Active Monitors range, meaning that room correction curves can be measured and written to the speakers themselves using the GLM Speaker Management Kit and GLM software. I was interested to see how closely ARC 4 replicated the curves that GLM came up with — and despite the fact that GLM only expects you to measure at a single point rather than 21 separate points, the answer was 'almost exactly'. There were only two areas of difference, both down to the ways in which the two systems operate rather than to measurement error. First, because the SAM system is aware of and can talk directly to all the speakers in a setup, it is able to directly handle bass management and also to phase-align the subwoofer with the satellites. This is not possible with ARC Studio, which has only left and right stereo outputs, not that the difference was very obvious in my room.

Second, in typically cautious Genelec style, GLM and the SAM system applies only subtractive EQ, and will not boost where there's a dip in the room response. There are sound reasons for this — boosting eats into headroom, and if you have a null due to a room mode, no amount of boosting will fix it — but, equally, there are times when being able to add a couple of dB somewhere in the spectrum can be beneficial. The curves that ARC Analysis came up with did feature small (as in 1 or 2 dB) boosts in the midrange, and consequently sounded a touch more assertive than the GLM correction. Neither was really better, and I quickly adapted to whichever I happened to be using at the time. The extra stages of A-D and D-A conversion introduced by adding the ARC Studio into the setup were not noticeable to my ears.

Not having used previous versions of the ARC software, I can't say how much better the new algorithm is, but I can report that the user experience is very slick. From analysing your room to choosing virtual monitors and custom responses, it's all completely intuitive and I never once felt the need to search for a manual. As for the ARC Studio hardware, it's designed as a no-frills plug-and-play box and it does exactly what it needs to, albeit with the millstone of that wall-wart PSU. It's by far the most affordable standalone hardware monitor correction system that I know of, and at the price it would be churlish to expect ribbons and bows. But at the same time, I do think there'd be demand for a more upmarket version with features like digital I/O, monitor control, bass management, a headphone amp and the ability to store and recall separate profiles for two sets of speakers. In fact, such is the pace of development at IK that I wouldn't be at all surprised to learn that something like that was already in the works. In the meantime, ARC Studio 4 does exactly what it's intended to, combining the independent, set-and-forget nature of a standalone hardware box with the flexibility of software. Truly, the best of both worlds. ■■■

The ARC Studio hardware is a no-frills box with analogue XLR ins and outs, a USB socket for communication with the ARC software, and a DC power input.

ALTERNATIVES

The obvious rival for ARC is **Sonarworks' SoundID Reference**. This can operate as a plug-in and as a systemwide app, and can also interact directly with some speakers and audio interfaces, but there's no current equivalent to the ARC Studio box. Other hardware solutions are typically much more expensive, much less user-friendly, or both...

$ ARC Studio $299.99; ARC 4 software and mic $199.99; ARC 4 software only $149.99.
W www.ikmultimedia.com

i73® PRO

The first ever USB-C Audio Interface with 73 style Class A preamps

Heritage Audio
The sound of yesterday built for tomorrow.

RAD
GLOBAL DISTRIBUTION
NEW YORK CHICAGO BATON ROUGE SHENZHEN PHUKET MADRID

New game... Classic rules

FULL ANALOG EXPERIENCE

Know more...

JOHN WALDEN

Westwood Instruments Lost Synth

Westwood re-imagine some classic instruments to take familiar sounds to new places.

Software Synthesizer

Virtual instruments come in many forms. Some might be classified as predominantly sample-based (for example, many acoustic drum or orchestral libraries). In contrast, others are synthesis-based (many software recreations of classic hardware synths fall into this type). Others, however, fall somewhere in between and Westwood Instruments' Lost Synth is an interesting example of that.

As a Kontakt-based instrument, it does have an underlying sample base, drawn from a collection of vintage synths including the Juno-60, Polysix and ARP Odyssey. However, while these samples undoubtedly shape the sound, it's the sound manipulation engine Westwood have built within Kontakt that defines what Lost Synth is really about. And, as that engine does — in parts at least — contain some rather unconventional elements, the sonic end result is also unconventional. If you like your synth sounds to be atmospheric, quirky, textural and possible with an added rhythmic element, Lost Synth might be right up your street.

Lost Synths Found

Designed for Kontakt 6.6.1 or later (free or paid version), Lost Synth features 80 underlying sample-based sounds. This comes in at a fairly compact 3.7GB in total, can be downloaded via Pulse and is authorised through NI's Native Access. The UI is nicely styled and the default Sounds page, as well as providing access to all of the 200+ Kontakt Snapshot presets, also hints at the twin layer nature of the sound design. As we will see in a minute, that's not quite the whole story, but the Sounds page is where you select sounds for the A and B slots from the underlying sample sources. It also provides a set of fairly conventional controls for activating each slot, setting level, pan, attack, release and two different tuning options. The large Blend knob adjusts the balance between the two sound slots and, if you activate the Motion option, this lets you automate the Blend based upon different LFO shapes, with sync to host or time-based speed control. MIDI Learn can also be used for hands-on control of any of Lost Synth's parameters.

If all you do is load one of the presets and tweak using the controls described so far, there is still plenty of sonic character to be explored. The presets category and sub-category labels hint at the somewhat leftfield sonics — Dirt, Dusk, Glow, Shrt (short for 'short and using shorter sounds) and Warp, for example — and while there are some conventional(ish) sounds within the collection (for example, within the '4ths, 5ths' category), that's not really what Lost Synth is about.

Engine Mechanics

That's where the three other pages of controls come in. The Processes page provides a selection of effects processing options including a full ADSR, Compressor, Overdrive, Filter (with LFO control), Chorus, Sample (bit depth and sample rate), Wow (pitch flutter) and a very effective Sub section. Most of these can be set to operate globally (the same settings are applied to both A and B sound slots) or independently for each sound slot.

The Places page offers various 'spatial' processing options with Ambience, Noise, Reverb and Delay. These are all well featured but it's worth noting the lo-fi-esque nature of the background sound elements you can add via the Ambience and Noise sections, and the options for some degraded delay effects. Add in some Wow from the Processes page, and things can get very nicely degraded and retro sounding. All of which then leads us back to the Sounds page because the Mood option found there essentially offers a number of preset configurations (Temper, Muse, Awe, Void, Blur and Yearn) of the Process and Places pages that you can blend into your selected sounds via the Level knob. It's like a macro-based multi-effects option and capable of totally transforming your sound in all sorts of unusual ways.

So far, so nicely quirky, but I've saved what I think is the most interesting element until last; the Memories page. The online documentation describes this as an arpeggiator and delay engine, but it serves those functions through a pretty unconventional control set. However, before getting into the controls, the key thing to note is that this page lets you add a third sound source into your overall preset. The underlying Shrt (short) sounds have been created with this page in mind, but all the different sound types can be used here. The

Westwood Instruments
Lost Synth
$139

PROS
- Capable of some wonderfully retro playable, textural and rhythmic sounds.
- Quirky UI with some great sound design options.

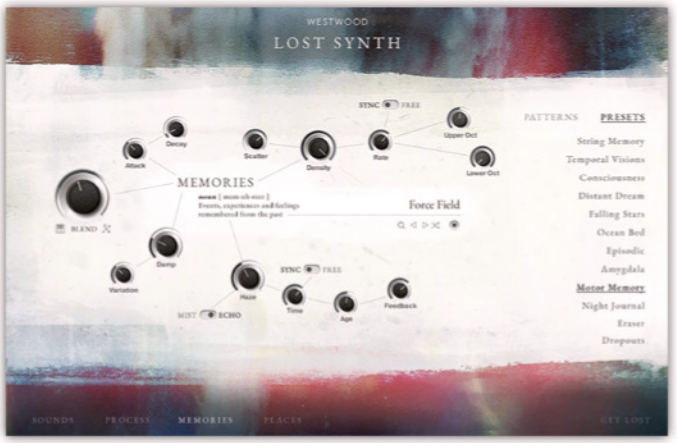
CONS
- Sonically, not for everyone, but very good at what it does.

SUMMARY
Lost Synth is a brilliant source of retro, dusty synth leads, textures, pads or rhythms housed within a wonderfully creative UI. A great option for media composers or ambient/textural electronica producers.

page includes both full Memories presets and Pattern presets (the latter seem to change the underlying arpeggiation patten).

The two branches of controls are both weird and wonderful and I'm still not sure I fully understand how they all interact. What I am sure of, however, is that the results can be very inspiring. You can tweak the attack/decay of the Memories sound component, add damping (reduces the high-end content) and random damping variations, use the Density control to adjust the amount of notes generated by the engine, adjust the sync of the arpeggiator to your host tempo, adjust the pitch range over which the arpeggiator generates notes, and then add a combination of Mist (something akin to delay and reverb) or Echo (a tape delay) with control over the time, depth, feedback and sync.

If you chose to add this third sound element, the additional Blend control on the Memories page then becomes an important part of your sound design. This adjusts the overall balance between what's generated via the Sounds page (from slots A and B) and the Memories page. Rotate fully left and you just hear the Sounds page, rotate fully right and you just hear the Memories page, or sit somewhere between (and adjust the setting via a suitable MIDI controller), and you can blend in just the right amount of the rhythmic element to your overall sound.

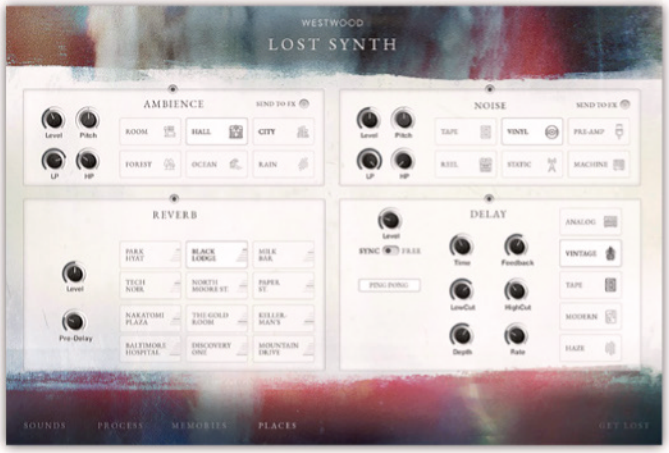


The Memories page provides a third sound source that adds a sense of rhythm to your sound with a very intriguing arpeggiator/delay style engine.

The final element of the engine is the Get Lost option located bottom-right of the UI. This provides access to the randomisation system that can generate a full preset at the click of the mouse. Rather sensibly, Westwood seem to have placed some useful constraints on some of the generated settings so that things don't get out of hand. As a result, this is capable of generating some very cool — and very usable — sounds. Just keep clicking and something inspiring will soon come along.

Lost In Sound

Once you have your head around the basic concept, the Lost Synth engine provides a super-intuitive performance platform, whether you want to create melodic parts, complex evolving textures, rhythmic patterns or some changing blend of all of these. The sounds themselves provide a starting point that undoubtedly has something of a retro feel, and you can take that lo-fi ethic some considerable distance further courtesy of the effects and processing options provided. I suspect that might mean that those looking for raging EDM synths could find more obvious choices elsewhere. However, if your music requires a more experimental element, Westwood Instruments have provided a really interesting sound palette and sound design engine for more ambient or textural electronica styles.



The effects options include some great options for 'degrading' your sound palette in some very cool ways, as shown here for the Places page.

The other obvious audience for Lost Synth will be media composers and I can see it being an excellent choice for the sorts of (often subtle) textural underscore that lots of modern drama, sci-fi, or horror (the bits before the gore actually happens) might require. Lost Synth's sounds would make it easy to create anything from the mystical or magical to the unsettling and unnerving, and a whole lot more besides. And, when you need to up the tension level, you gradually dial in that rhythmic element to add a pulse or a sense of time running out.

Conclusions

This is the first product from Westwood that I've had the chance to explore, and I suspect they may be a new name to many *SOS* readers. However, I have to say that I'm very impressed. Lost Synth is cool, quirky, relatively compact and delivers some fabulous sound design options in a (mostly) intuitive UI. Yes, the sounds favour a more experimental type of music creation using retro, dusty, tones but, within that musical ballpark, this really is very good.

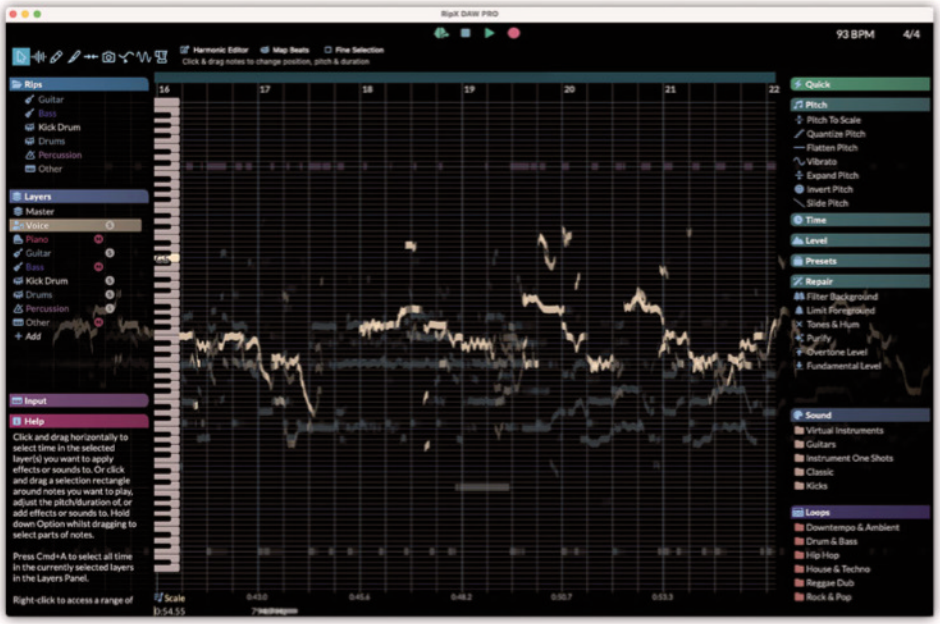
While not in the pocket money range, it's very sensibly and competitively priced. Even if Westwood are new to you, I'd encourage media composers and electronic musicians with a love of textural/ambient styles to check out the audio demos available on company's website. They many be all you need to take the plunge and get lost in a little Lost Synth exploration.

$ $139
W www.westwoodinstruments.com

Hit'n'Mix RipX DAW PRO

Source Separation & Audio Processing Software

RipX PRO offers high-quality stem separation and an intriguing suite of tools for audio editing.



JOHN WALDEN

We've taken a couple of dips into Hit'n'Mix's RipX in recent years (see the April 2023 and September 2021 reviews), but AI moves rapidly and Hit'n'Mix have now released v7 of RipX. This release brings a more straightforward dual version approach with RipX DAW and — with some additional advanced options — RipX DAW PRO. The core functionality of earlier versions remains intact but, of course, the latest releases bring refinements and new features. Let's explore...

When Is A DAW Not A DAW?

For those new to RipX, a brief discussion on terminology is important to place the product in context. Digital Audio Workstation is a very broad term. The most common type of software referred to as a DAW tends to be audio and MIDI recording and mixing applications such as Pro Tools, Cubase, Logic, Reaper, DP and the like. However, there are other types of software-based environments for working with digital audio that offer different sorts of functionality and are designed for different types of tasks. Audio editing environments such as WaveLab or SpectraLayers (both by Steinberg) or iZotope's RX would be obvious examples; all three let you work with digital audio, but they are DAWs that focus on audio editing tasks.

RipX DAW offers audio editing and elements of music production and/or creation, so it certainly fits in the broad category defined by the term DAW. However, in the same way that WaveLab, SpectraLayers or RX provide workflow and functionality different to that found in Cubase or Logic, so does RipX. Indeed, in the world of DAWs (in that broad sense of the term), given both the workflow and feature set, RipX is somewhat unique. To misquote a well-known line from Mr Spock, it's a DAW, Jim, but not as we know it...

Fix It In The Unmix

RipX first caught general attention for its ability to separate a stereo source file into a number of instrument-based layers (stems). When you drop a suitable audio file into the Rips panel, a dialogue lets you choose which stems you wish to extract, with Voice, Bass, Drums/Percussion, Guitar, Piano and Other available as options. While there are a number of tools that can now perform this type of stem separation task very well (including SpectraLayers and RX), the quality of the separation processing within RipX has always been a highlight of the software. It remains so in this release and Hit'n'Mix have continued to refine the process further, including some noticeable gains in the speed with which the 'ripping' process is performed.

I re-ripped a few commercial tracks that I'd ripped with the previous release and, even without dipping into the PRO version's Audioshop toolset (which, as mentioned below, offers some options for further cleaning up the stem separation process), I found some noticeable and worthwhile improvements in the quality of the stems produced. For example, the isolated vocal layers seemed a little cleaner. Yes, harmony vocal parts and vocals originally mixed with more obvious ambience treatments (reverb and delay) can still make life difficult, but there did seem to be fewer unwanted sonic details finding their way into the vocal layer. Whatever your use case for the output of the stem unmixing functionality, cleaner separation means less subsequent editing work and a faster workflow, so improvements on this front will always be welcome.

For many potential users, the stem unmixing capability may still be the headline attraction of RipX regardless of the other functionality Hit'n'Mix have added over more recent release cycles. If creating remixes or musical mashups is your end

Hit'n'Mix RipX DAW PRO
$198

PROS
- Stem separation that's as good as it gets.
- Some truly intriguing audio editing and music creation possibilities.

CONS
- Some may find RipX's unique approach presents a steep learning curve.

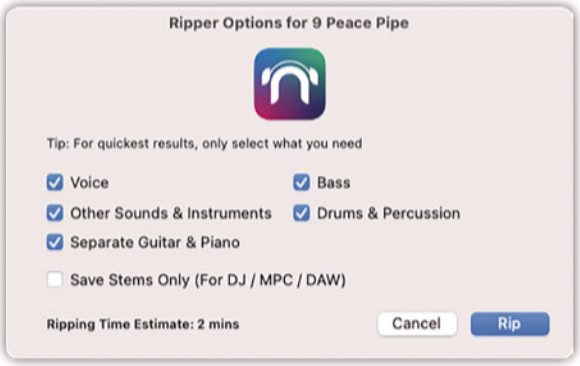
SUMMARY
RipX DAW offers class-leading stem separation and intriguing audio editing/manipulation options, accessible via a somewhat unique workflow.

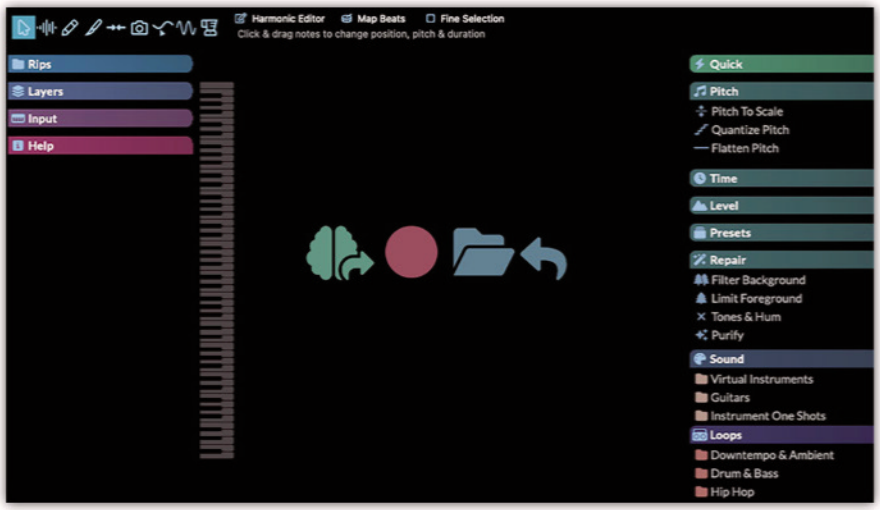
goal, then the price of entry for RipX DAW may well be justified for this feature alone. However, the same capability is also an incredibly powerful educational tool, especially as it is so easy to modify the playback tempo independently of the pitch. Whether it's to create a backing track so you can practice your vocal cover, or slow the tempo down on an isolated guitar stem to work out the impossibly fast solo, RipX's rips are useful for much more than grabbing an a cappella vocal.

My (AI) Generation

While there would appear to be plenty of clever AI within the stem separation algorithms, this release sees Hit'n'Mix leaning into AI within a further element of RipX. Clicking on the new green 'brain' icon within the main screen takes you to a dedicated Hit'n'Mix web page that contains links to AI-based music generation platforms. The idea is that the user might generate an initial musical starting point using one of these platforms and then rip it within RipX. Once broken down into stems, you can experiment with any or all of RipX's audio or MIDI-based editing and manipulation tools to transform the AI generated original and craft it into your own musical idea.

Hit'n'Mix indicate this resource list will be regularly updated as the available options evolve but, at the time of writing, the most accessible of the services listed was Stable Audio. This site offers both free and paid subscriptions, and the music generated is based upon whatever text prompt (and a few other user-defined parameters) you wish to enter. While I'm not sure the full tracks generated by Stable Audio are going to put composers out of work yet, where I did find this effective was in asking the AI to generate specific parts of a musical track. For example, I prompted it to create a lo-fi track featuring just drums, bass and acoustic piano, rather than a full arrangement. When this was imported into RipX and broken into stems, it was easy to see how the individual elements could then be manipulated, the sounds tweaked or replaced, and MIDI generated. The result was a cool little loop that could easily serve as a seed for further work, whether within RipX itself or exported out to the likes of Cubase or Logic.





The green 'brain' icon provides inspiration courtesy of AI music generation.

Going PRO

As mentioned earlier, the PRO version adds some additional functionality. This includes options for separately cleaning up noise within unpitched sounds, harmonic editing, a RipScript language for building your own tools, the Repair panel tools and the Audioshop toolset (opened from the Panels menu). These options offer a number of additional corrective and creative possibilities for more advanced users.

Particularly interesting are the tools within the Repair panel. Whether you're working on all the material within a layer, or pitched and unpitched elements individually, the Audioshop tools let you select either component and then provide options for repair and cleaning. For example, the Filter Background option lets you set a dB threshold to filter out quieter audio elements within the selection. Purify massages out amplitude changes for a smoother sound. The Tones & Hum control lets you remove unwanted sound elements based upon a user-defined bandwidth. If you need to really extract the last few percentage points of quality out of that isolated vocal layer, then there are tools here that will let you attempt that.

As well as quality improvements, the stem separation process is now more efficient.

It's also worth noting that the PRO version provides various ways to integrate it with a more conventional DAW, be that as an external editor or via the VST3, ARA2 and AU RipLink plug-in.

The X Factor

To return to where we started, RipX is a DAW but, given the eclectic and unusual combination of audio editing and sound manipulations tools, it is something of a unique product within that broad class of application. So, given its somewhat unusual nature, who might fall into Hit'n'Mix's target audience?

First, if your primary need is stem separation, RipX DAW — in its standard or PRO versions — remains impressive, capable of standing its ground against any of the obvious competition. Second, if your preferred music creation process involves lots of sample manipulation, or creating new musical ideas from loops, the RipX DAW method provides an intriguing, unconventional, and somewhat unique way to explore that.

In either of these contexts, RipX DAW is not really like any other audio application out there and, for the more advanced editing options, there is a learning curve to be climbed through hands-on experience. Sensibly, Hit'n'Mix do let you try before you buy: there is a 21-day trial available on their website and, if you have any sort of experimental nature, it's most certainly worth experiencing.

$ RipX DAW $99, RipX DAW PRO $198.
W www.hitnmix.com

Electro-Harmonix Pico Triboro Bridge

Overdrive Pedal

Most drive pedals are based around traditional analogue circuitry but in this Pico-series pedal, EHX have used digital technology to coax three different characters of drive from a single pedal. Not only that but they have also included powerful tone-shaping controls, making this a very versatile little pedal that's able to cover everything from the merest hint of drive to full-on fuzz. Being digital, it takes more current than its analogue counterparts, but a PSU is included and if you plan to use your own pedalboard PSU, 100mA of current is required.

The three modes, selected using the Type button and flagged by the tri-colour LED, are Overdrive, Distortion, and Fuzz. As with the other pedals in the range, the colours are green, orange and red, and the orange and red can look quite similar. There are the usual drive and volume controls, which are common to all modes, plus two tone controls for bass and treble, though in Fuzz mode the leftmost knob controls a noise gate while the right-hand knob becomes an overall tone control, with a slightly resonant low-pass characteristic.

Overdrive goes from low to mid gain and has an open voicing that lets most of the character of the instrument come through without change. To my ears, when using a guitar with single-coil pickups and with the pedal plugged into a clean amplifier, this sounded just slightly aggressive with the tone controls set flat, but backing off both the bass and treble helped smooth things out. With the amp set to already add a bit of hair to the sound, the overdrive pushed it into blues territory, easily maintaining a natural tonal character.

Distortion adds more gain and gets you into classic rock territory and, again, the tone controls can be used to refine the sound and to dial back any unwanted edginess that might creep in. Those tone controls, which have a Baxandall characteristic in this mode, have a lot of range. Switch to fuzz mode and the bass knob controls a noise gate threshold, while the treble knob allows for extreme tonal variations, ranging from thin and buzzy to very fat and soupy. It can deliver a Big Muff kind of tone but at brighter settings does that raspy 'Satisfaction' thing pretty well too.

There's also a secret Input Contouring EQ setting, activated by pressing and holding the Type button, which causes the LED to flash rapidly eight times. This shifts from an unfiltered input to one that tames the low end while boosting the mids, a characteristic that many will recognise from Tube Screamer-type pedals. The effect is fairly subtle but to my ears smooths out a little of the overdrive/distortion edginess I commented upon earlier and using it I could get very close to the sound I get from my own Fulltone OCD pedal. The input EQ setting is remembered on power up/down so you can pick which mode works best for your particular type of guitar pickup and stick with it.

To summarise, this is a very versatile and compact pedal with plenty of tonal flexibility, though if you want to switch from one mode to another, be prepared to adjust the controls each time you do so.

Paul White

$ $144.40.
W www.ehx.com

Rode NT1 Signature

Cardioid Capacitor Microphone

Rode introduced the original NT1 way back in 1991, and it's been so successful that last year they released the fifth generation of this mic. Indeed, when I'm asked by musicians or aspiring engineers to advise on a first 'proper mic on a budget', it's pretty much always been on my shortlist. I was recently sent the new NT1 Signature for review — this one is an all-analogue affair with the usual XLR connector on the bottom, but it's worth noting that there's also a USB version with 32-bit converters and some DSP on board (it also includes the XLR connector), and if you're interested in reading about that one, check out Sam Inglis' March 2023 review: www.soundonsound.com/reviews/rode-nt1-5th-gen.

As well as a smart-looking mic (available in silver or black), you get some nice accessories, and I was impressed with the quality of the included shockmount, which has a neat attachment for fitting the included pop filter. There's an XLR cable included too, so you only need a mic stand and preamp/audio interface and you're good to go.

Like its predecessors this NT1 is a cardioid-only capacitor mic that features no additional pads or filters. But the lack of such 'bells and whistles' is by no means a sign of cheap quality. While the biggest evolution in the fifth-generation NT1 is arguably the digital connectivity, there's plenty more to commend this analogue-only model too. For example, unlike many budget-friendly mic companies, Rode design and build their mic capsules in-house, and the capsule is a truly one-sided design, rather than the common dual-sided affair with the rear diaphragm disconnected. And, as Sam pointed out, the quoted self-noise of 4dBA is very impressive. It should make the NT1 Signature a great option for dialogue and podcasts — since the voice is exposed a lot of the time, it can make you really appreciate a mic that doesn't add unwanted noise!

I run a commercial recording studio, in which I record a large range of vocalists and instruments. If I'm evaluating high-end mics, then, after a short period of testing to ensure they're capable of delivering the goods for paying clients, I tend to throw them straight into use on real sessions. With more budget-friendly options such as the NT1 I adopt a slightly different approach — not because I think the mic won't be up to the job, but rather because many clients probably have an NT1 at home and in this setting they want/expect to see an expensive mic in front of them! Instead, I'll rig the mic alongside my studio's more established mics. This is a great way of broadly assessing the merits of a mic like this.

»

ADAM AUDIO

GENAUIGKEIT!

Means “accuracy” in German.

THE ADAM AUDIO A SERIES → ADAM-AUDIO.COM/A-SERIES

» So how did the NT1 Signature fare on these sessions? The short answer is that when compared with mics that often cost as much as 10 times the price of the Rode, the NT1 held its own very well. On male vocals, in fact, I would have been quite happy to use the NT1 for the projects I was working on. I could hear some small differences in the lower registers of a voice, and the midrange felt a little 'pinched' on louder sections when compared to an expensive valve mic I was using, but in general the sound was good and there was certainly nothing problematic. I could hear a whisker more difference on female vocals: the NT1 felt slightly less 'silky' than my usual options, but again, I was sitting there specifically listening for small differences — it's not the kind of thing that would hold anyone back if recording their music with the NT1.

When it came to instrument recording, I managed to try the mic out on a few different acoustic guitars, on an upright piano and as a mono drum ambience mic, and in each setting the NT1 Signature captured an accurate and nicely balanced picture of what was in front of it. In particular, when recording some delicate fingerpicked guitar acoustic guitar, the mic's low self-noise was evident — I was able to crank my preamp's gain and capture all the detail I wanted.

As with its predecessors, then, the NT1 Signature is a good-quality condenser mic, and while it offers no bells and whistles in terms of features, it does offer exceptional value for money. The fact you get both a good shockmount and a pop filter included in the price is icing on that cake. I've described some of the small differences that I could hear when comparing it with some very expensive mics that I use daily, but those differences are nothing like as great as you might assume, and as long as you have a decent audio interface and a basic grasp of mic technique, the NT1 is definitely capable of professional-level recordings. It's a great option for musicians who are beginning their journey into recording, but for small studios looking to pad out their existing mic collection this traditional 'starter mic' could be a surprisingly cost-effective choice too. *Neil Rogers*

$ $159.
W www.rode.com

Universal Audio UAFX Brigade

Chorus & Vibrato Pedal

UA's digitally modelled Boss CE-1 analogue Chorus Ensemble was transplanted from their plug-in portfolio back into hardware in their Astra pedal, but it reappears now as part of the UAFX compact range. The name, Brigade, is retained in the compact pedal and references the charge-coupled devices, popularly referred to as 'bucket-brigade' chips, used in the original circuit for the analogue delay line that was modulated to produce the chorus effect.

The CE-1 pedal has the honour of being the very first unit to bear the Boss name, and made the distinctive modulation effect popularised by the Roland JC-120 Jazz Chorus amp available in standalone form for the first time. The thing that made the JC-120 combo's chorus a bit special was that it was rendered as a 'wet-dry' effect, using a separate power amp for each of its two speakers. The CE-1 pedal retained that faux stereo with separate wet and dry outputs and would reward you with a similarly spacious effect when hooked up to two amps, but for many users, even just the mono output was exciting enough in the late '70s.

Popular with keyboard players, guitarists and the odd bass player, the CE-1's chorus had just a single control, labelled Intensity, with the separate Depth and Rate controls operating only on the footswitchable Vibrato effect — this was a pitch-modulated output with no dry signal mixed in. The input impedance of the CE-1 was a bit low for passive guitars, so it helped to have a buffering pedal in front of it, and it was always hard to avoid a creating a bit of crunch from the preamp, to the point where it really just seemed like part of the effect. Brigade gives you the option to include or omit the distortion and coloration of the preamp when the pedal is in bypass, but it is of course always on when the effect is active. Anyone who had a CE-1 will remember it as a frustratingly noisy pedal. In contrast, the Brigade is delightfully noise free, even when you turn up the Level control.

True to form, UA have modelled the sound of the CE-1 with uncanny accuracy to my ears, whilst also replicating the limited control functionality. In Chorus mode, Depth has the function of the CE-1's Intensity control, setting both modulation rate and, I think, slightly changing the wet/dry mix. All the classic chorus sounds are to be found in about a 30-degree arc in the middle of the rotation — too low and there's not much happening, too high and you are into full-on 'seasick' warble. In the middle, though, it's just perfect! I could never get the separate Rate and Depth controls of the CE-1's 'improved' successor CE-2 and similar pedals to sound like this. Perhaps the limitations of the CE-1's preamp are a useful factor, plus the fact that later chorus pedal designs tended to employ longer delay times and more 'voices', whereas the CE-1's delay time is barely out of flanger territory and modulates just a single 'voice'. It may just be my associating it with so many classic tracks, but for all its limitations I find that something about this sound always just sits perfectly in the context of a mix. Or maybe that's because there's really only one setting that works so you are not temped to try anything 'out of the zone'!

It's mono in and out, there's no Bluetooth or app to worry about and the USB-C is just for firmware updates, so you just set the controls and go. But it's still a digital pedal, and whilst the odd millisecond or two of latency when the effect is active won't matter much, there's one scenario in which it does. If you want to replicate the CE-1s faux stereo, perhaps by using the Brigade on a splitter or mixer send, your direct dry signal won't be time-aligned with the dry element included within the mixed effect output, so you may notice that the sound is very slightly thinned out compared to either channel on its own. But it is still well worth doing, so long as you can pan the channels apart or send them to different amps. The effect is far more subtle and spacious than in mono. The full wet/dry faux stereo CE-1 experience is available in the Astra pedal.

Of course, you can get a brand-new, analogue CE-1 from Boss themselves, in the form of one of the modes on a CE-2w. Is it more accurate? I'm not sure we can say what 'accurate' is anymore, with most of the original CE-1s having drifted out of spec or been modified, 'improved' or repaired. Either way, UA's Brigade does exactly what I expect a CE-1-type effect to do, and in a compact format with top-mounted jacks, too. What's not to like? *Dave Lockwood.*

$ $199.
W www.uaudio.com

Seven kits recorded by iconic producer George Massenburg in a 330 square meter room, a wealth of sound shaping tools, effects and options – Superior Drummer 3 is nothing short of a production powerhouse. If you like mixing and tweaking drums, discover our award-winning software today.

WWW.TOONTRACK.COM

TOONTRACK

MODULAR

Knob Technology SGR1806-20

Eurorack Module

Some 50,000 light-years away is Magnetar, a weird neutron star of immense magnetic density that identifies as a soft gamma repeater. Just over there in my rack is its modular namesake, a weird module of immense signal density that identifies as an analogue drum synthesizer. One can emit gamma rays that disturb the curvature of space-time, and the other can take CV and bang out beats that will get even the most uptight aliens dancing. Welcome to SGR1806-20, the Eurorack module.

The SGR1806-20 (let's call it SGR for the sake of simplicity) is an analogue drum synth module from Knob Technology. It's brash, spacey and corrupted, pulling you away from any sense of order into a warped mass of unstable waveforms, strangled noise and torrents of unexplained extraterrestrial communications. It lurches from clicks to grit to getting pulled through gravel and on to the sort of sonic mayhem that sounds like you're tuning the radio on a dying spaceship. If there's something stable in here, then I haven't found it, but that's probably the point.

The SGR has two sources of sound: a 'Voice' block with a clash of three triangle VCOs, and a 'Noise' block containing three noise generators. These get mixed, folded and distorted to arrive at a VCA as a space-time anomaly.

The Voice is built from three unsynchronised triangle waveforms. It's inspired by the Buchla 259 complex oscillator and there are definitely complicated things going on between these oscillators. There's a Spread function that detunes oscillators one and three with reference to oscillator two. They go in different directions depending on which way you turn it and at no point will they bring it all back to some lovely resolution, it's always just a little bit off.

Two other controls force the voice into self-modulation. FM pushes each oscillator into the next whereas Feedback pipes some of the output into the voltage summing unit of the 1V/oct input. The result is chaotic with occasional moments of clarity. With everything dialled back you can get SG to play a tune, but frankly, it's not that interested. What it seems to be

looking for is explosions of energetic texture, and those are very easy to find.

The Noise engine is ridiculous. It has a single control that sweeps it from white noise to fax machine, broken radio to system crash.

Lastly, we have a four-stage wavefolder and distortion. The wavefolder plays with each voice differently. The Voice gets bent, folded and generates harmonics while the Noise gets phase-shifted and together they "create new spectra at the intersection of filtering and distortion". I don't know about that but it's certainly true that a little bit of folding enlivens the signals. The Distortion rounds off this sonic adventure with a suitable dollop of overload.

As a drone I thought it was like some kind of energetic alien space radio searching for life in the far reaches of the cosmos. However, that's not what this is about. The protagonist of this story is the Envelope and the rupturing influence provided by the Trigger input.

The envelope is a straightforward percussive decay envelope that ranges from snappy clicks to open infinity. Through the array of yellow buttons, the envelope can be pumped into pretty much everything. It instantly takes any parameter to its peak and then drags it back to Earth. If you consider how the FM engine is feeding oscillators to each other and the Spread is speeding up or slowing down alternate waveforms, or how the fold is disrupting the shape, and the Noise Tone is still searching for Alien Classic FM, then that envelope can do an awful lot of damage.

Feed it some triggers and it starts to spit and revolve, pulsate, crack and squelch its way through rhythms. And as with most percussion synth voices the magic happens when you patch in some modulation. Everything has a CV input so you can flip the mix knob between booming kick drums and frazzled spurts of noise, or tickle the decay from penetrating clicks into zaps and warbles of crashing harmonics. On the down side I thought the front panel was a bit of a mess, difficult to read and not exactly easy on the eye, but it did light up in interesting ways. It took quite a bit of experimentation to find my way around, and the results were often unexpected and maniacal. One thing I found quite hilarious was that I could be crafting away on an intricate cascade of interesting clicks only to find a universe of extraordinary alien sounds when I opened up the envelope. But with the right combination of triggers, envelope routing and modulation it was a totally magnetic experience.

SGR1806-20 is capable of conjuring up an endless supply of broken rhythms and angry textures. It's thoroughly weird and satisfyingly alien. *Robin Vincent*

$ €340

W trianglecore.rocks

Xaoc Devices Ostrawa & Bohumin

Eurorack Module

Aren't the full names of Xaoc Devices' modules just so soothing? Unboxing and mounting the Ostrawa and Bohumin I'm reminded of the Polish developer's generous ability to make one feel like their system might be edging just a little closer to the fabled European electronic music studios of yore. I favourably reviewed the Sofia Transcendent Waveform Analog Oscillator: Model Of 1955 a while

back, and this time around it's the turn of the Ostrawa Full Stereo Voltage Controlled Mixing Console: Model Of 1966 and the Bohumin Mixing Console Commander: Model Of 1966. You may have picked up on the geographical theme; the city of Ostrawa, by the way, is in the western Czech Republic, adjacent to which lies the town of Bohumin. That should give you a good indication about the relationship between these two modules.

The Ostrawa mixer offers four stereo channels of mixing and a single stereo aux send. It's a very nicely laid-out thing, with each channel offering a volume knob, a clickless mute switch, pan pot, aux send attenuator and LED level meter. Below these are per-channel stereo inputs and CV inputs for panning (or Balance in stereo) and volume, meaning that any of the Ostrawa's DC-coupled channels can act as a CV-controlled VCA if desired. The lowermost row of jacks is allocated to the stereo send and return, sum outputs and also a useful pair of Direct Input jacks for chaining a submix into the equation, or — as I often do — for feeding a module with its own volume control into the mixer without having to use up a channel.

It's a nifty and compact design that fits a lot onto the faceplate without feeling cramped. I do like the tactility of sliders, but knobs were the right choice here and suit live performance well since they're well clear of the mixer's patch points. While it's fair to say that four channels isn't masses, there's a generous amount of I/O to speak of if you really need some workarounds. I've never been averse to using two mono signals through a stereo channel, for instance, or to using the aux return as an extra input. The aux send — a huge selling point for those whose workflow is anything like mine — can be pre- or post-fader, switchable simply with a long press on the mute button and indicated by the button LED changing from green to orange.

The Bohumin expander contributes an additional aux send, B, though this is far from all it adds to the Ostrawa. It also furnishes both sends A and B with their own master return attenuators and provides jacks (here labelled Active) to mute and unmute any of the mixer channels with gates. I was a little disappointed to find aux B can only operate post-VCA, but it makes up for this limitation (at least in part) by offering CV control over each channel's send, which is not possible with aux A and opens up a host of interesting patching potential, particularly when used in conjunction with the Active inputs and the VCA inputs on the mixer proper.

The Ostrawa is a sister module to another Czech-themed mixer, the Praga (no prizes for knowing where that city is), which is similar in architecture and almost identical in its layout but differs in a number of functions. One thing about Xaoc's range is that it does very well to build an ecosystem of modules that don't just work with one another in the conventional modular sense, but in many cases expand one another's capabilities from within. The Ostrawa is a prime example: it can not only be expanded with the Bohumin via a ribbon cable, it can also be chained to one or more Pragas or other Ostrawas for a mega mixer that can occupy the entirety of your lower 3U if you want it to. The Praga has its own expander to boot, the 10HP Hrad (meaning 'castle', with Prague containing the country's most famous), which ostensibly endows its parent module with a master section including a headphone output.

That's an astonishing amount of mix-and-match flexibility on offer here, and I'd argue that while these are on the pricey side in the first instance, in the longer term it all amounts to something quite budget-friendly, since you can expand your channel count slowly as your needs grow. I for one would be very interested to see Xaoc Devices consider releasing modules of single channel strips for channel-by-channel customisability, anchored by the Hrad's master bus... I digress. I've tested a number of Eurorack mixers, and the Ostrawa/Bohumin team is up there with the very best of them.

William Stokes

$ Ostrawa $549.99, Bohumin $289.99.

W www.xaocdevices.com

Qu-Bit Electronix Mojave

Eurorack Module

The Mojave is concerned with all things granular. Conceptually, of course, this means sand. This 'granular sandstorm' is named after the Mojave desert, "drawing its inspiration from vast swaths of desert in the American Southwest," and sure enough its faceplate conjures a rather lovely light-up graphic of sand dunes and a zephyr. The image is a good one, in the sense that granular processing can be quite hard to rationalise. The allegory, I suppose, would be that Mojave renders your source audio a structure made of sand, or grains, and then presents a means of controlling the wind blowing that sand around. This could entail throwing caution to the wind (geddit) or zooming right into the micro-sound domain to rearrange things with the most precise of breaths.

The host of parameters for rhythmic and melodic manipulation makes a rather crowded faceplate whose controls could easily populate a panel twice its size.

A central Rate knob controls the frequency of grain generation, and can be clock-sync'ed. Drift is concerned with where in the source audio the Mojave draws its grains from, and at extreme settings will span the buffer to grab grains at random. This often works in tandem with the Zone control, which determines the audio buffer position. Distribute generates more and more complex rhythm events over the course of its travel distance, while Structure deals in pitch and scale. Whirl sends grains bouncing, or drifting, around the stereo field. There's also a Speed parameter to control pitch (which I was very happy to find can track at one Volt per octave), two different types of Freeze function and even an end-of-chain effect, named Gust, for adding internal feedback or even reverb. Finally (just to turn things on their head all over again) there's an onboard MEMS microphone, so the Mojave can take in acoustic audio as well. This has limited applicability — and would be essentially unusable in any environment but a quiet studio — but fair to say it seems to have been included as more of a bonus feature than a core component, and what a bonus it is!

Qu-Bit are certainly a wildly ambitious bunch, and beyond panel graphics to die for — which actually do contribute to workflow, incredibly — they also love a poetic motif or two. Take the Mojave's scale quantise button for example, which cycles through blue, green, yellow or purple indicators for different scales; only here it's Sky Mode, and cycles through 'Dawn', 'Day', 'Dusk' and 'Twilight'. Romantic. The grain generation mode button specifies where the Mojave gets its instruction to generate grains from: the clock, the input signal amplitude or manual triggers. Or, in Qu-Bit's terms, 'Erode', 'Shear' or 'Chisel'. Being a Brit, it's customary for me to scoff a little at this sort of thing, but in reality it simply suggests significant attention to detail and a great deal of pride in the design — something that should only ever be lauded.

Even the simplest of input signals can lead to gorgeous results from the Mojave. I started off feeding it the most basic of drones, which I was soon spattering around my stereo field with percussive, ratcheting complex rhythms, subtly shifting pitch to add chorus or ricocheting it between the extremities of various scales. Other sounds, for instance my own voice into the microphone, yielded much more complex results, and I particularly enjoyed sending two different signals into the left and right inputs to be processed together. Drums are endowed with complex syncopation and timbral depth, while live keys or even guitar can be sent into far-flung rhythmic and tonal territories.

The world of granular seems to have opened up considerably in recent years. This is likely thanks to developers' increasingly inventive explorations of digital platforms, which I partly ascribe to the synthesis world's move away from the fetishism of all things analogue, but that's another discussion. I still love analogue, by the way. There are other modules out there dealing capably in this world, for instance Instruō's Arbhar, but Qu-Bit have come upon something quite special here, and a good deal cheaper than the Arbhar, it must be said. It also gives the Make Noise Morphagene, which is a different beast but certainly operates in a convergent world, a run for its money — particularly since it can process pitch at 1V/octave. From wild explorative gestures to imbuing sounds with gentle movement, the Mojave is a formidable tool. Highly recommended. *William Stokes*

$ $399
W www.qubitelectronix.com

What's New

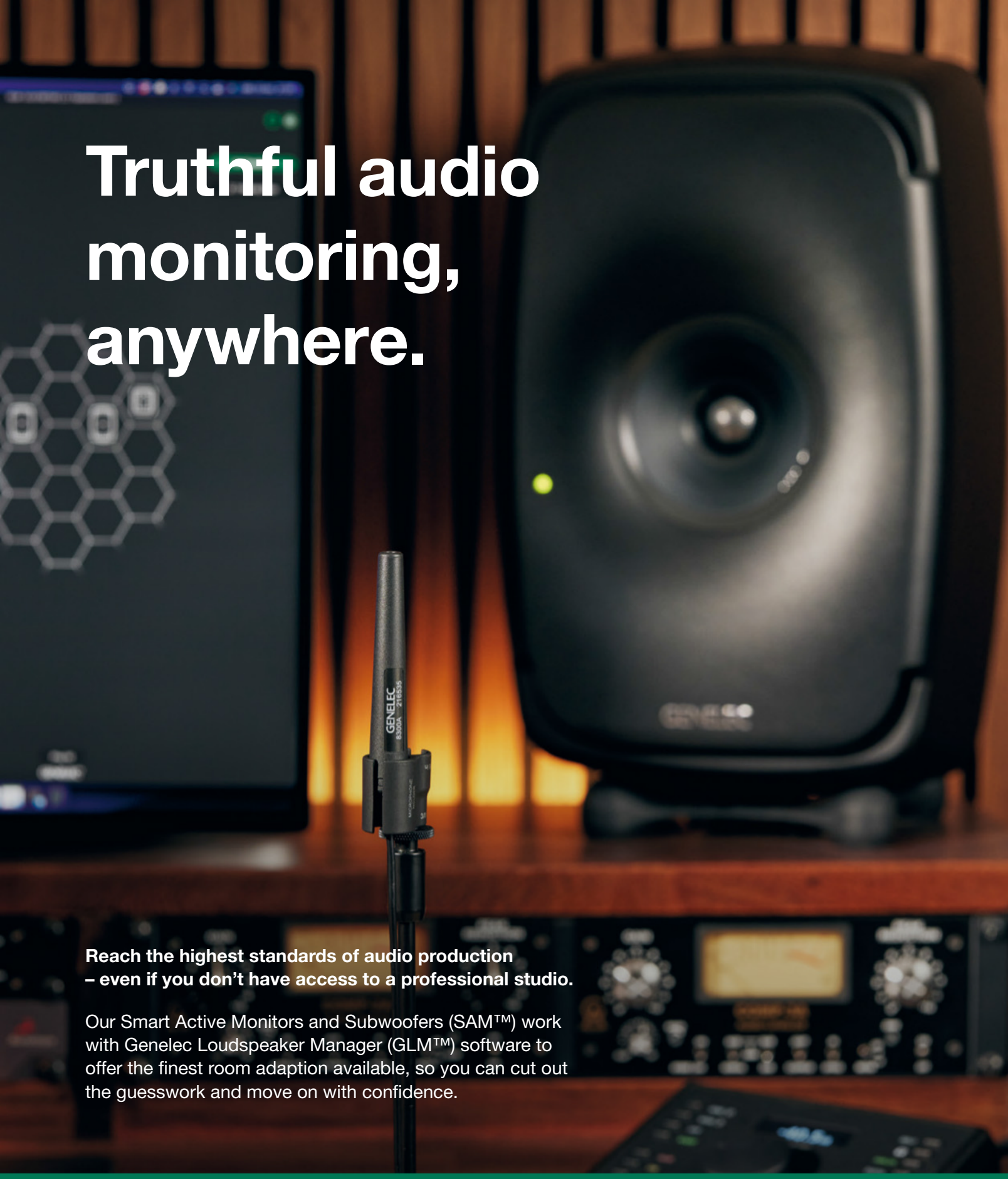
Describing themselves as "a collective of experienced, specialist synth-builders and designers", the **Glasgow Synth Guild** exists to "bring new electronic instrument designs into the world, and re-imagined reissues from some forgotten relics." The collective's first offering is the Oct Tōne, an eight-step control voltage and pulse-signal sequencer in 10HP. It's not quite an Instruō Module as its look would have you believe, but it is a reboot of an early design by Instruō founder Jason Lim. Look out for a review in a future issue. www.glasgowsynthguild.com

Further south, **ALM/Busy Circuits** have unveiled the CIZZLE, a dual digital 'phase distortion' oscillator in 16HP. The module, say ALM, is inspired by the classic Casio CZ series synths; drawing upon its distinctive phase distortion with primary and secondary oscillator layering and detuning, 'morphable' phase distortion wave generation, ring modulation and noise. www.busycircuits.com

Noise Engineering have unveiled the newest addition to their 6HP Legio platform: the Sinc Legio is a compact stereo oscillator full of attitude, boasting wave morphing, wavefolding, phase modulation and more. It joins a line-up with the Roucha Legio filter, Tymp Legio percussion module, Librae Legio dynamics processor and more — all of which have interchangeable firmware. Just as we were going to press NE also announced the Opp Ned, a CV-controllable four-channel arpeggiator with editable patterns. Talk about indefatigable! www.noiseengineering.us *William Stokes*

Technique: Getting Wet

Effects, particularly spatial effects, constitute a relatively modern facet of modular synthesis, but this need not mean they should only be used as enhancement for existing sounds created on other modules. Many Eurorack mixers have very useful effects sends, but this can mean that reverb and delay can become consigned to the 'sprinkling' category, and as such be under-utilised. It's very much possible to make effects an intrinsic part of your patch, bedding them into a tapestry of texture and movement. One patch I've been enjoying recently starts with a fairly basic sequenced oscillator, fed through a simple, pleasant reverb. I've then been treating the reverb as a sound in and of itself, sending it to a totally separate channel on my mixer. Try sending your reverb or delay through a VCA; experiment with controlling the VCA with different wave shapes to create unnatural swells of space around your central pattern. After the VCA, patch your reverb through a filter; patch a second sequencer (or any other stepped CV) to control the filter cutoff with the resonance cranked up, then try using skipped steps to create cross-rhythms to interact with your central pattern. You could then try distorting your verb, modulating it at audio rate — the possibilities are endless. In no time you'll be creating all manner of intricacies out of the simplest of sequences, blowing the notion of what reverb can be wide open! *William Stokes*



Visit genelec.com/studio-monitors

GENELEC®

MIX RESCUE

Trevor Piggott

The foundation of a good, engaging mix is a strong arrangement. But how far can you go to improve things at the mixing stage?

MIKE SENIOR

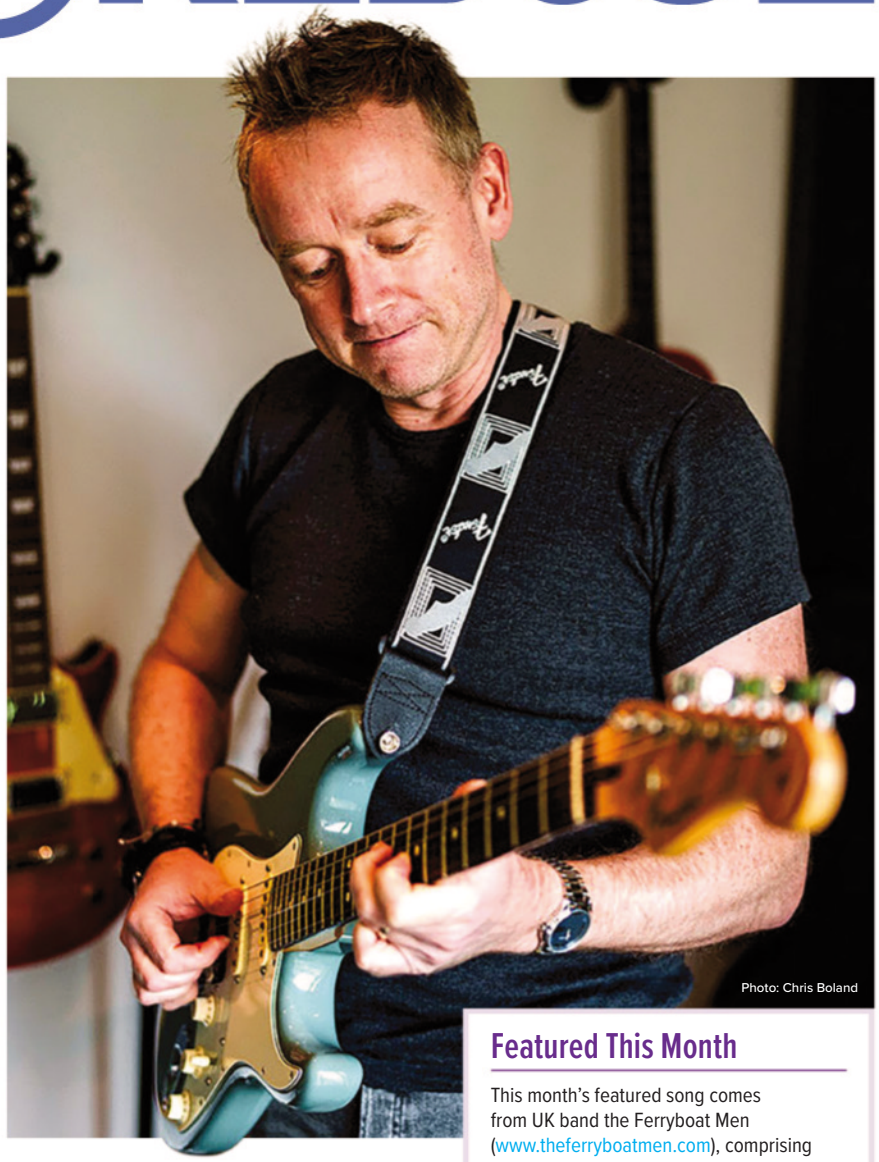


Photo: Chris Boland

When *SOS* reader Trevor Piggott recently sent me over a mix he was struggling with, I could hear that the sonics lacked some clarity and punch compared with the Bob Clearmountain mix he was referencing against (the Simple Minds song 'I Wish You Were Here'), and that the lead vocal wasn't commanding the listener's attention enough. There was, however, another more insidious malaise, because he'd also fallen into a trap that ensnares many project-studio users: relying too heavily on repetition, especially of the copy-paste variety.

It's an easy thing to fall into, and often goes like this. First you create a four-bar pattern with maybe drums, bass, and some chords, and then you quickly copy-paste that so you can crack on with writing a song over it. By the time you're done, you're beginning to get a bit bored with the bare-bones arrangement you've heard looped so many times, so you begin adding more parts to freshen up the pattern. But while each new layer does re-enthuse you at first, its novelty inevitably declines with repetition as you work, eventually leaving you with an arrangement that, despite being saturated with musical parts, leaves you with a niggling sense at mixdown that something's still missing — no matter what processing or effects gizmos you try.

In this article, I'd like to share some of the practical arrangement and mixing techniques I typically use to address such issues, and show how I used them to rework Trevor's production, upgrade the mix sonics, and reinvigorate his enthusiasm for the song.

Once Less With Feeling

One of the first things I did with Trevor's song was look for opportunities to shorten the structure. After all, if any kind of musical pattern gets staler the more you repeat it, why not simply reduce the number of repeats? As it happened, there was an eight-bar instrumental section that was treading water between the second chorus and the onset of the guitar solo, so removing that was an easy

Featured This Month

This month's featured song comes from UK band the Ferryboat Men (www.theferryboatmen.com), comprising Trevor Piggott (www.trevorpiggott.com) on guitar, keys, and vocals, and Stephen Hurren on drums and percussion. Having first jammed together at school, both guys have had varied musical careers since. Trevor has toured with several different rock and folk bands, branching out from there into songwriting and scoring work for film and TV. Stephen has also toured extensively in support of Arista Records artists such as the Chester Project, and has recently been working the festival circuit with his own Back To The 50s trio. The inspiration for this particular song is the tragic 11th Century story about the unrequited love of Juliet Tewesly — whose ghost apparently haunts the band's local pub!

One easy way to differentiate the sections in your arrangement and provide more of a sense of build-up through your song is to reserve some sonic layers for later in the timeline — as you can see Mike doing here with some of the chorus backing-vocal tracks in his remix.

win. But there were also two separate intro sections that delayed the arrival of the first vocal verse until the 30-second mark. I chose to sort of fold those into each other, to get to the lyrics 10 seconds sooner. On a smaller scale, I also pruned out a few repeated sections on a per-track basis, muting the bass guitar during the introduction, progressively weeding out more backing-vocal layers for the earlier choruses, and removing the chorus piano hook from the guitar solo section (where the backing track was plenty busy already).

Simple cuts like that will only take you so far, though. Another more useful strategy is to modify some of the repetitions so they sustain the listener's interest better. If you think about it, exact repetition is actually quite an unnatural thing in real-life music-making, because human musicians never really play the same thing twice — indeed, that's an inherent part of the magic of the live gig experience! So in this case, when I realised that the main piano riff comprised three repetitions of the same rhythm, I decided to edit the second one into a slightly different shape. Similarly, the chorus' main lead-vocal melody comprised a pair of identical lines which I was able to transform into a slightly more interesting 'call and response' pairing by shifting the final note of the second line to a higher pitch. A melodic clean guitar riff that looped through the pre-choruses and choruses was edited so that the pre-chorus version became much sparser and simpler, which meant that the chorus iteration then felt like a musical development, instead of a straight copy. And there were multiple instances where I was able to mute individual instruments for a moment just to kind of remind the listener of their presence — most notably the drum kit just before verse one and during the first half of verse two, and the bass guitar just before the final choruses.

> "Don't use the same widening tactic for everything, because every widening effect has its own potentially undesirable side-effects."

And, of course, if you go to the trouble of capturing live performances, it's a shame to squander their humanity by subsequently looping sections of them — which is what the band had felt obliged to do here with their live drum tracks, because of difficulties maintaining the song's swung groove against the click during recording. Fortunately, they'd archived the original live take, so I was able to re-import that into my mix session and use editing to deal with its timing issues directly instead — this meant I could retain all of the player's nice little musical accents and pattern refinements. And, while I was at it, I decided to pare back the drum kit texture during the first 40 seconds of the song by muting the room mics and tom close mics and low-pass filtering the kick-drum and snare close mics, thereby giving more scope for the drums to build up through the timeline.

Stealth Layers & Arrangement Build-up

Where regular editing methods don't provide enough scope for introducing variation, a great alternative can be adding 'stealth' layers to supplement the existing parts. I used this tactic for the echo-y triplet guitar chords that underpin most of this song, EQ'ing a stock Jazz Guitar patch from NI Kontakt to get a similar sound, then programming two different upper layers to subtly differentiate the part's verse, pre-chorus and chorus voicings. (I also reused the region-specific low-pass filtering dodge I'd tried on the drums, restricting the guitar's upper spectrum early in the song to improve the long-term dynamics.) The programmed bass part benefited from some layering too, with an added sub-bass synth lending the line extra power for the later choruses and during the second half of the guitar solo.

You have to be careful when adding extra parts at mixdown like this, because few musicians like to feel that you're tampering with the essential musical material. So it's wise to keep such contributions in the 'subtle to subliminal' range. But if you feel that a section of the arrangement needs something a bit »

more ostentatious, you can reduce the risk of a negative response if you create that new element by remodelling some existing recorded track, perhaps from a different section of the song. For example, I'd jettisoned a piano special-effect track while whittling down the song's introduction, but later I was able to use this as a 'new' atmospheric element to differentiate the second verse and pre-chorus from the first.

This trick helped with the song's second chorus too. You see, the third chorus had been bolstered with heavily distorted electric guitars, but there was no real sonic progression between choruses one and two. Luckily, Trevor had recorded DI signals for those parts, so I could generate less heavily driven versions of the guitars during the second chorus, bridging the 'energy gap' between the first and third. Similarly, moving an iteration of a clean guitar riff from the first part of the solo to the start of chorus five not only generated an arrangement 'lift' for the second half of the solo, but also for the second of the final choruses.

Difficulties with the click track during the tracking sessions had led the band to build their original drum track from short loops of the drummer's playing, but this robbed the song of both short-term musical variations and long-term performance dynamics. To remedy this, Mike first reconstituted and synchronised the original continuous drum track using detailed edits, and then enhanced the section differentiation by thinning the track count at strategic moments.

Mixing For Clarity

If you ask me, it's hardly worth doing any real mixing until you've adequately addressed arrangement issues like these. What's the point in getting bogged down in plug-in settings before you're able to judge sounds within their final context? So it was only once I'd resolved my repetition concerns that I turned my attention towards my first main mixing goal: achieving 'clarity', in other words making sure all the layered parts could be heard without the overall mix tonality becoming woolly or bloated.

There are lots of fancy ways to attack this but it's important not to neglect simpler tools. Straightforward filtering is a great workhorse, for example, and it had an important role to play in this mix. High-pass filters on the drum overheads, room mics, electric guitars, piano and effect returns really helped keep the low end clutter-free for the kick drum and bass guitar, while low-pass filtering helped remove abrasive upper-spectrum masking frequencies from the distorted guitars, as well as pushing some of the harmony vocals and clean-guitar riffs more into the background behind the (more musically important) lead vocal and solo guitar parts.

Another family of 'clarity enhancement' techniques essentially involves moving energy from overpopulated bits of mix real estate to more sparsely occupied regions. As with a lot of project-studio multitracks, this project had an overabundance of lower midrange, a good chunk of which was coming from the bass guitar. So I cut that firmly around 140Hz, but then compensated for that loss of energy by adding more true low end (from the added sub synth layer) and boosting the midrange around 1.5kHz. To put it another way, I traded some low midrange (that the mix had too much of) for more sub bass and midrange (where the mix had greater headroom available). Likewise, there were two different tracks (the piano atmospherics in the second verse/pre-chorus and the clean guitar riff first heard underneath the second half of the guitar solo) where I deliberately mixed in some octave-upwards pitch-shifting to move their frequency emphasis further up the spectrum and hence reduce their reliance on the low mids.

One way of apparently introducing variation into a repeating part is to subtly layer a similar-sounding MIDI instrument alongside. In this remix, for example, Mike used a Jazz Guitar patch from Native Instruments' Kontakt to extend the upper harmony voicing of the song's main echo-y guitar loop, to suit different sections of the arrangement.

This principle of shifting between areas of the mix can apply to the stereo image too, and I used a few different methods to clear space at the centre of the stereo image for the most important arrangement elements (ie. the kick, bass, snare, and lead vocal). For mono tracks, panning is usually the first port of call, and I did hard-pan the most heavily distorted rhythm guitars in this mix. But for stereo channels (such as the Hammond organ and the clean guitar riff subgroup) the simplest way of clearing the centre is to reduce the level of the stereo signal's Middle component using Mid-Sides (M-S) processing. Occasionally, you may need frequency-selective control, and I did here: I wanted to thin out just the lower spectrum at the centre of the echo-y guitar's stereo image, and in such cases an M-S equaliser will likely be more suitable.

To clear the central image of an instrument, an alternative to cutting its Mid component is to find a way to boost the Sides, and then turn the whole track down — either way you're shifting the Mid-Sides balance in favour of the Sides component. There are lots of ways to do this, from specialist insert processors (such as the freeware Polyverse Wider plug-in I applied to Trevor's chorus piano hook) to send effects (such as the widescreen modulated delay/reverb send effect I used to spread the clean guitar riff and piano atmospherics). My main advice here is that you don't use the same widening tactic for everything, because every widening effect has its own potentially undesirable side-effects »

If there's one note in a musical part that dominates over the others, that may prevent you from fading that channel up enough to hear the other notes with sufficient clarity. One solution to this problem is to use a specialised multi-notch equaliser (such as the Voxengo GlissEQ plug-in Mike applied in his remix) to rebalance the errant note so that all the notes can come through more clearly.

» (perhaps it makes the timbre chorus-y in mono, or it distances instruments from the listener), and I think it pays not to compound them.

It may seem counterintuitive, but the imbalance between an instrument's different notes can also cause a loss of mix clarity. For example, the echo-y guitar part in this mix featured one overplayed note that overwhelmed the mix before any of the instrument's other notes were coming through clearly. With MIDI parts, some swift reprogramming can remedy this, but with audio I often find that surgically notching out some of the offending note's harmonics can achieve a similar result, making it possible to raise the instrument's fader to a point where the rest of the notes become more audible. You can implement this kind of processing with most regular digital EQ designs, but I use it often enough myself that I appreciate those few EQ plug-ins that offer a special 'multi-notch' filter type that you tune to the target note's fundamental frequency and then it automatically cuts a specified number of that note's harmonics into the bargain — Voxengo's Gliss EQ, for example, or Melda's freeware MEqualiser.

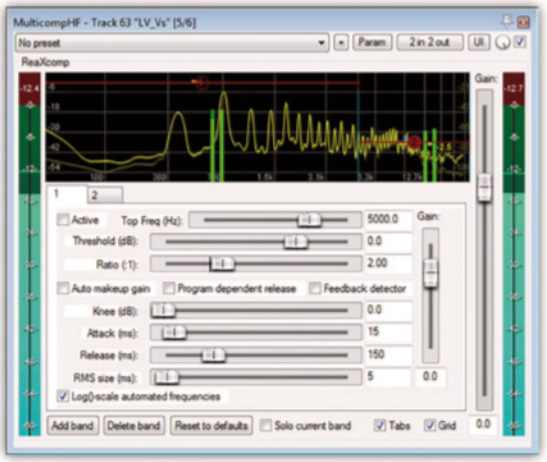
Again, it's easy to get carried away with more complicated processing hacks like these and forget that probably the best all-purpose clarity-boosting trick is simply riding the channel fader! It might seem a bit low-tech to just push up a track's fader when it's worth hearing and pull it down again when other tracks are more important, but it's a tried and true technique — and I used it all over this remix, especially on the guitar and piano.

Powering Up The Drums

Another important part of this remix was trying to increase the power and punch of the drums, which is something I'm often asked for tips about. To state the blindingly obvious, the first thing to do is turn them up! The rub is that fading up any instrument in a mix often shines an unwelcome spotlight on performance inconsistencies that need sorting out. For example, I realised from comparing Trevor's mix against the Simple Minds reference that the kick and snare needed greater prominence in the balance, but neither of those drums felt consistent enough in level that I could simply push up the faders — some hits always ended up feeling too strong or not strong enough. So one of the first things I did was limit them both. Limiting might seem an extreme choice, but I wasn't triggering more than 4dB of gain reduction and I set a fairly long 450ms release time, so the processing was really just levelling out the performance hit-to-hit rather than reshaping the drum envelopes.

Sharpening the attack of your drums can also help them cut through more, and most DAWs now have transient processors that are great for this. In this case, I used one built into my Reaper DAW for the kick drum. In some situations, though, the frequency-selective control provided by some third-party plug-ins can be handy. When I applied full-band transient processing to the snare in this mix, it seemed to add more of a sharp 'edge' than a meaty 'punch' (if you'll forgive the wine-tasting terms...), so I was happy that I could turn to iZotope Neutron's multiband transient processor to boost the drum's sub-200Hz region independently.

Upper-spectrum EQ boosts are frequently used to emphasise the forwardness and aggression of drums »



Increasing the density and consistency of a vocal signal's upper spectrum can help bring it to the front of the mix, and Mike achieved that for the verse vocal in this mix using a combination of analogue-modelled saturation (to generate additional harmonic content) and fast-acting compression acting on the frequency range above 5kHz.

GO BEYOND...

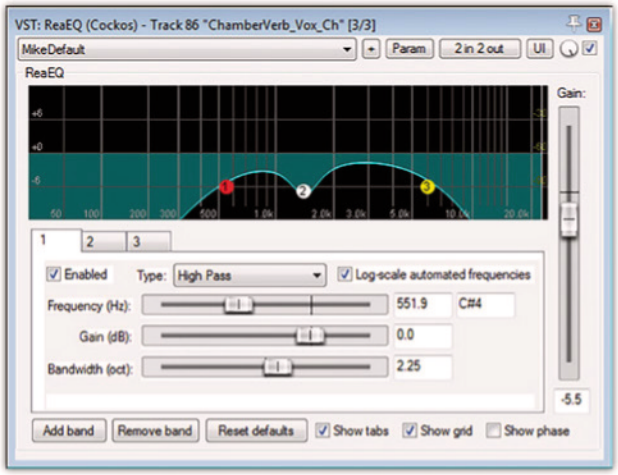
WINNER
SOS AWARDS 2024

The H90 Harmonizer® has the power to take you beyond the effect horizon into unchartered sonic territory. Packed with 62 effect algorithms, including groundbreaking effects like Polyphony, Prism Shift, and Wormhole, the H90 is a powerhouse. Capable of running two effects at once (with spillover), it features an intuitive user interface designed with players in mind. The H90 is Eventide's Next Step and is destined to sit at the heart of your rig for years to come.

Eventide®

eventideaudio.com

Eventide is a registered trademark of Eventide Inc. Harmonizer is Eventide's trademark for a special effects device incorporating pitch change. © 2022 Eventide Inc.



Here you can see three things Mike did to the vocal reverb in his remix to keep the lead singer sounding up-front: de-essing the reverb send to avoid consonant sounds 'splashing' in the effect tail; adding 50ms of pre-delay to the reverb itself; and assertively EQ'ing the reverb return to distance the effect and reduce its midrange content.

» (I boosted 9dB on the kick drum, for instance), but it's important to realise that EQ can also undesirably soften your drum transients. The reason is that most normal EQ designs don't just boost and cut frequency regions. They also delay (or 'phase-shift') some of them as a side-effect, which can effectively 'smear' well-defined transients, making them sound less punchy. In practice, phase-shift side-effects won't cause appreciable transient-smearing problems most of the time, as long as you keep your EQ moves fairly moderate. But there's one common EQ move that you need to be wary of in this respect: high-pass filtering. This is something people often do almost as a reflex in small studios, to avoid problems with subsonic rubbish that doesn't show up on typical two-way nearfield monitors. But with drums, high-pass filtering can significantly reduce the solidity and power of the instrument's attack on account of the phase-shift, even if the frequency response effects seem pretty minimal. I deliberately didn't high-pass filter the snare for this reason.

The subjective power of drums isn't just about their front-end 'spike', though, because there are other ways you can contribute to the illusion. Adding some dense room reverb is a time-honoured approach, and hearing this in the Simple Minds mix encouraged me to try something similar on Trevor's snare. With any heavier reverb like this, it's important to realise that the effect will to some extent be perceived as part of the overall snare tone, so do make sure you spend enough time auditioning different reverb patches, and don't be afraid to use those effects assertively to get a combined timbre that really fits the mix. In my case, I spent a good 15 minutes trying different reverb plug-ins and presets before settling on a blend of two different custom-tweaked room patches. Even then, I didn't leave those reverbs static throughout the song, and automated their return levels to 'inflate' the snare sound more during the higher-energy sections of the song.

Emphasising the sustain of your drums can also suggest to the listener that the kit sounds more powerful, and compression is the natural choice for this. The danger is that traditional compression carries the risk of counter-productively blunting your drum transients. This is one of the classic scenarios where parallel compression (mixing compressed and uncompressed drum kit sounds together) really comes into its own, because the transients of the uncompressed signal will always reach the mix bus, irrespective of how sadistically you drive the compressor on the parallel path. In this remix, for example, I cranked up some serious gain reduction using an aggressive fast-attack, fast-release compression setting based on the 'all buttons' mode of a UREI 1176 limiter. This all but flattened out the transients of the compressed signal, but the uncompressed channel's transients still arrived safely at the mix bus. The result: more sustain, without any loss of attack.

Leading From The Front

My final key requirement for this mix was that the lead vocal should be right up front where it would demand the listener's attention for the lyrics. As with the kick and snare, that meant both increasing its overall level and controlling its balance more stringently, to avoid it ever overwhelming the

backing track and undermining the band's sense of size. A single layer of dynamics control is rarely enough to walk this mixing tightrope, so I first used a limiter to catch the loudest peaks, then followed it with slower-acting 3:1 ratio compression, to squeeze the overall dynamic range by 3-4 dB.

Beyond basic dynamics processing, there are a number of things you can do to bring vocals to the front. Anything that increases the number of harmonics in the upper spectrum usually helps, for example. Not only does it brighten the tone, but that sense of brightness makes the part less susceptible to frequency masking. In this case, I set up a dedicated parallel distortion channel for this and, for instance, mixed in the distortion's added harmonics to supplement the singer's natural high frequencies during the verses. I also used a multiband compressor for all the song's lead vocals, to further brighten the frequency region above 5kHz without overemphasising the already bright-sounding noise consonants and breaths.

If you're using any vocal reverb effects, there's always a danger they'll drag your singer backwards in the depth perspective, and judging by Trevor's mix and choice of reference material, I knew I'd need to take precautions against this, because vocal reverb was definitely on the menu! So here's what I did:

1. I fed some of the reverb from a tempo-sync'ed feedback delay effect. This makes it easier to lengthen the apparent reverb tails without as much of a sense that you're washing out the whole mix with cavernous reverb.

2. I made sure that all the reverbs I used had at least 10ms of pre-delay. This helps separate the dry vocal signal from the reverb, weakening the distancing effect.

3. I de-essed the reverb sends, to avoid splashy consonants in the reverb return from pushing in front of the dry vocal signal.

4. I used EQ to cut regions from the vocal reverb returns wherever they seemed to muddy the vocal tone, become too audible in the midrange, or sound too bright by comparison with the dry signal. A useful tip is to temporarily turn the reverb up 3-4 dB too loud while EQ'ing, because that makes it easier to hear which specific frequencies need cutting.

Remix Reactions

Trevor Piggott: "Now that is a GREAT mix! The difference is night and day. I'm struck by the remarkable clarity of each individual element, with the cleaner electric guitar tones helping with the separation and the drums delivering a more impactful punch. The lead vocals have been propelled forward relative to the backing vocals and harmonies, drawing attention to the lead melody, and the guitar solo has also acquired enhanced grit and prominence. I've had a tendency to hold back on my vocal levels and guitar solos, but this mix has emphasised the importance and effectiveness of spotlighting lead lines.

"The overall precision of the arrangement is more evident too, and I love how Mike's created light and shade and dynamics. I particularly liked the strategic entry of drums in the second verse, which added an extra layer of intrigue and dynamic flair to the track. My own mix all sounds very samey and dull by comparison — just goes to show how important mixing and arrangement is! This song is the prelude to an album of similar tracks, so I'm sincerely grateful for the benefit of Mike's expertise in setting a benchmark for the entire record."

Photo: Chris Boland

Listen To & Remix This Track!

You can find a selection of audio examples relating to this remix, as well as downloads of Trevor's raw multitrack files and Mike's completed DAW project, at https://sosm.ag/mix-rescue-0524.

Besides the vocals, the other lead part was a guitar solo, which I also wanted to have at the front of the mix. Initially, though, the recording presented two difficulties. First, the distortion felt like it had been driven a bit too hard, such that I didn't feel I could fade up the more musically important pitched information without my ears being fatigued by aggressive upper-midrange hash. And when I used upper-midrange EQ cuts to tackle the harshness, the guitar ended up feeling distant compared with the cymbals and other guitars in the arrangement. Second, a long modulated echo effect had been baked into the guitar recording, so I couldn't adjust those effects independently.

Fortunately, Trevor had recorded a DI signal for this part too, which gave me some room to manoeuvre. (It's never a bad idea to capture a 'safety' DI signal when you record electric guitars, even if you never actually use it — it can really save your bacon at mixdown if you misjudge the amp or stompbox settings.) Re-amping the DI with a less heavily-driven sound allowed me to reintroduce some less abrasive-sounding high frequencies into the mix. This already helped pull the guitar sound forward. Then, because this re-amped signal included no effects, I could adjust the wet/dry mix of the echo effect according to how I mixed the re-amp with Trevor's original guitar part. I *could* have just ditched the original guitar effects entirely and created totally new ones for the re-amp track, but the echo had a certain character that I felt might prove tricky to recreate from scratch — I risked throwing the baby out with the bathwater!

Music Before Mix

In this month's remix, I've showcased lots of different mixing techniques for enhancing clarity, beefing up your drum sound, and bringing lead lines closer to the listener, but none of those will do you much good if your listener loses interest in the music. So it pays to think twice every time you're tempted to just copy and paste.

ENJOY THE VU!

After 85 years of active service, the humble VU meter remains as useful as ever in today's digital studios — despite BBC engineers nicknaming it 'virtually useless'!

HUGH ROBJOHNS

As I was listening to David Mellor's *SOS* Podcasts about gain-staging (www.soundonsound.com/author/david-mellor) recently, my ears pricked up at his passing comment that "BBC engineers refer to the VU meter as virtually useless." It wasn't a surprise, exactly, as I was told exactly that during my initial BBC training in the early 1980s and I believed it for years. But I came to understand that the claim was, in fact, based on a fundamental misunderstanding of how the VU meter was designed to be used!

My view today — and it's shared by many professional mastering and recording engineers all around the world — is that the VU meter remains very useful, even in today's digital studios. So, in this article, I'll take you through the virtues and practical benefits of the octogenarian VU meter, and explain how the BBC got it so wrong. To do that, we need to start with a little history...

Origins: PPMs & SVIs

The VU meter was conceived in 1939, through a collaboration between research company Bell Labs and the American broadcasters CBS and NBC, and a paper they published in 1940 described what they called the Standard Volume Indicator (SVI). As the SVI meter's scale was calibrated in 'volume units' (and marked 'VU'), it became known popularly as the VU meter — in much the same way that the modern BS.1770 Integrated Loudness meter is often called simply an 'LUFS meter'. It's testament to the genius of those 1940s engineers that the SVI's core specification lives on today, virtually unchanged, as *IEC 60268-17:1990 Sound System Equipment. Part 17: Standard Volume Indicators*.

Of course, various audio meters were already available in the 1930s, so why exactly did CBS, NBC and Bell Labs feel the need to design their own? Well, in the late 1930s, radio broadcasting was a rapidly expanding business globally, and there was a requirement to monitor and control the broadcast programme levels consistently and reliably. Over in Europe, many national broadcasters had developed their own metering systems and nearly all were Peak Programme Meters (PPMs or, more accurately, QPPMs — see the 'Quasi-PPM' box for more on this). To protect the transmitters from overload, European engineers believed it was necessary to monitor the peak programme levels, and their work resulted in the DIN (German), Nordic (Scandinavian) and BBC (British) PPM meters. Although these each had different scales they all employed similarly complicated, valve-based active electronics to detect and display pseudo-peak levels.

Such complex metering systems, though, were relatively expensive to manufacture, so although the American broadcasters were aware of the European designs they thought them impractical for use across the vast US radio industry. A simpler, more affordable, passive solution was desired that, preferably, would be capable of indicating the average signal level in a way that reflected how listeners would perceive the volume.

Hence their SVI, and this meter comprised just three passive elements: an adjustable attenuator, a copper-oxide rectifier and a bespoke moving-coil meter, built to detailed specifications in terms of its sensitivity and needle ballistics. The rectifier was required to convert an AC audio signal into a DC voltage that could move the meter, and the passive

A New Standard Volume Indicator and Reference Level*

H. A. CHINN, MEMBER, I.R.E., D. K. GANNETT, MEMBER, I.R.E., AND R. M. MORRIS, ASSOCIATE, I.R.E.

The SVI meter — the first to be scaled in Volume Units — was described in a paper published in 1940, and its core specification remains in use today.

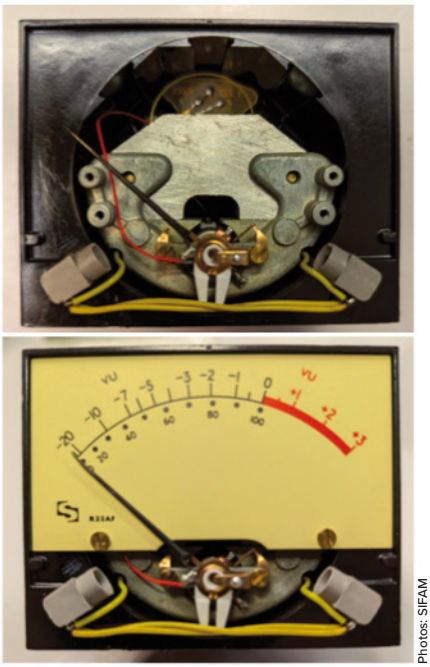
copper-oxide type they specified avoided the expensive valves employed in European PPMs, as well as the associated power supply. The bespoke moving-coil meter's relatively slow (300ms) needle rise and fall times ensured the meter would register low-frequency and sustained sounds better than brief transient peaks, so it correlated reasonably closely to perceived audio volume, hence the decision to use the Volume Unit scale. By far the most complex and expensive element was the constant-impedance variable attenuator, which had to maintain a consistent 600Ω load across the line being metered.

Back in the 1930s (and well into the '70s), professional audio interfacing used a matched-impedance format rather than the matched-voltage one of today, but the variable attenuator was an essential element to this meter. It adjusted the moving-coil meter's native sensitivity to accurately assess the signal level present on the line being monitored. Today, we expect metering to indicate the signal level directly: if you look at a conventional sample-peak bar-graph in your DAW, you can see whether the signal is peaking at 0dBFS or -10dBFS, or whatever. But the SVI was designed to be used differently, and this explains its relatively limited and cramped scale range. In practice, the SVI was wired across the signal line to be monitored, and the attenuator was adjusted until the meter hovered just below the 0VU mark (nearly 100% modulation). It was actually the resulting attenuator setting rather than the meter display itself that informed the user of the nominal signal level. The SVI's attenuator covered levels from 0 to +24 dBm, and that made it suitable both for standard studio lines (which typically operated at +8dBm) as well as telecoms lines (which operated at +16dBm).

Although often overlooked, both the SVI meter and the VU meters that evolved from it have two scales. The primary calibration, with which we're all no doubt familiar, is marked in decibels relative to 0VU. But there's a secondary scale beneath that shows the programme modulation level between 0 and 100%. 'Modulation' is a term rooted in AM radio broadcasting, and it refers to the strength of the audio signal being broadcast: 0% modulation means that the carrier is present but conveying no audio signal, while 100% modulation means it's carrying as much audio amplitude as is possible without overload. On an SVI/VU meter, the 100% modulation level aligns with 0VU (this is coincidental; it relates to the meter's slow ballistics) and 0% is a little below the -20VU mark. In use, a steady 1kHz or 440Hz tone at the desired Operating Level (whatever that may be) should read 0VU, while a varying audio programme should stay below the 100% mark most of the time.

Studio Operating Level

Another important aspect of the VU is its default alignment level. With the original SVI's variable attenuator set to 0dBm (ie. no attenuation applied), a steady tone of +4dBm at the input (yes, it's dBm rather than dBu because this was designed for a 600Ω environment!) gave an indication of 0VU on the meter, and so this is the reference level for the moving-coil meter itself. The SVI was widely employed in broadcasting and, later, in American music studios, and it was calibrated to the relevant



Photos: SIFAM

A glimpse inside SIFAM's old MkIV VU meter. Note that, as with all VU meters, there are two scales on the front: one in Volume Units, and the other showing modulation from 0 to 100%.

system operating level, whatever that happened to be. By the 1960s and '70s, though, multitrack recording was becoming popular and the associated equipment needed lots and lots of audio meters for the mixing consoles and the tape machines. The PPM was far too expensive to meet that need, but a simpler, more affordable version of the SVI meter might — if only that expensive attenuator could be omitted. So that's what the manufacturers did. Dropping the attenuator meant the meter sensitivity couldn't (easily) be adjusted, but it was thought not to matter in the context they would be used: if you wanted to record »

» 'hot' you just had to put up with the meter being pinned to the right.

As I've explained, the basic meter's inherent sensitivity gave a 0VU reading for an input level of +4dBm. This became the *de facto* console operating level simply because it was convenient. As American recording and signal processing equipment spread around the world with VU meters, so too did this 'standard' +4dBm studio operating level. By the end of the 1970s, the 600Ω matched-impedance interconnecting format had faded away, to be replaced with matched-voltage interfacing. But the same reference voltage (1.228V RMS) remained, so today there's an almost universal association between 0VU on the meter and a +4dBu standard operating level.

The Virtually Useless BBC!

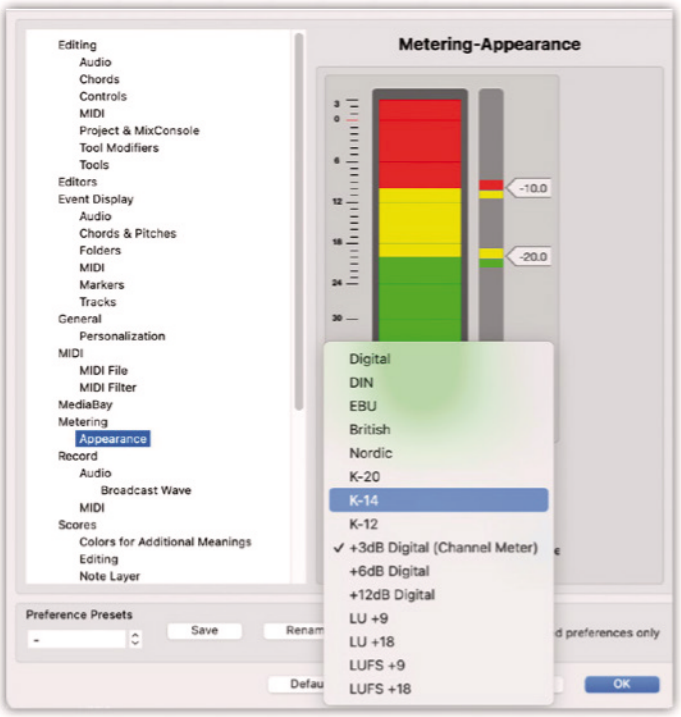
In a parallel universe, given a different design of moving-coil meter, 0VU could just as easily have been specified for a signal level of 1V RMS or -10dBV, or any other entirely arbitrary value determined

Metering & Headroom

In the days of analogue audio systems, all of the available metering (VU, PPM, Nordic, DIN etc.) inherently only provided a view through a small window of the audio system's total dynamic range. This was intentional, and encouraged users to optimise signal levels around the operating level highlighted in the meter, which essentially hid the available safety headroom margin. When Sony introduced their first professional digital audio recording systems, the PCM-1600 and 1610 CD mastering processors, their intended operating level was indicated by a '0' on the bar-graph meters, with 20dB of headroom above, exactly in line with analogue practices. But, unlike analogue meters, the full headroom margin was actually shown because the use of pre-emphasis made it depressingly easy to accidentally overload the system!

Oddly, when the improved PCM-1630 model was launched, Sony re-scaled the digital bar-graph meter to have '0' at the top (the clipping level). The intended operating level was then indicated with a user-adjustable LED marker which could be set anywhere between -10 and -20 dBFS. Every digital device ever since has used that same digital metering paradigm, showing the entire headroom margin with a vast metered dynamic range, yet without any explicit operating level reference point! So it's no wonder that users tended to peak close to 0dBFS, with all the associated hassles: the meter scaling has inherently encouraged them to do so. On more modern digital devices with sample peak meters that support personalised meter colours, I always configure my meters to show green up to -20 or -18 dBFS, yellow from there to -10dBFS, and red above -10dBFS, and it's remarkable how such a simple scheme automatically encourages correctly optimised recording levels with decent headroom margins.

Finally, it's interesting to note that with the passage into 32-bit floating-point recording, the available digital meters (which still stop at 0dBFS) no longer show the considerable (safety) headroom margin hidden above. Just like ye olde analogue meters! What goes around, comes around...



Some DAWs now allow you to change the channel meters to use scales that leave you some headroom, but where sample-peak meters are the only option it can help to set your meter colours to make it obvious when a signal is peaking too high.

The Sony PCM-1630 was the first digital device to align the meter's 'zero point' with digital full scale — and since then, digital meters have tended not to have headroom built in.

Photo: Akakage1962 (Creative Commons 3.0)

purely by the mechanics of an available moving-coil meter. And this is a subtle, yet critically important distinction that lies at the heart of the BBC's misunderstanding.

A lot of engineers still mistakenly believe that 0VU is defined as +4dBu, but this is absolutely not the case! 0VU does equate to +4dBu in most commercial studios and in equipment designed for use in that environment, but that's only because it was an easy alignment to adopt, given the native sensitivity of the VU meters originally employed in such equipment. 0VU is actually defined as the system Operating Level, whatever that may be for the equipment and system being metered. For example, 0VU is often aligned at -2dBu in parts of France, to 0dBu in most European broadcast companies, to -20dBFS in a SMPTE-calibrated digital system, and to -18dBFS in an EBU-calibrated digital system. All of these are perfectly legitimate.

Fortunately, this wide variety of VU calibration standards is easily accommodated because almost all modern VU meter implementations employ active electronics to drive the VU meter — in effect, the original SVI's passive matched-impedance attenuator has been replaced by active adjustable buffer circuitry that makes adjusting the meter's sensitivity to suit any desired Operating Level very straightforward in most cases.

Now that you're armed with an understanding of how 0VU is supposed to be aligned to the local Operating Level, consider what would happen if it were mis-calibrated 4dB too high. That's exactly what the BBC did, institutionally, when accepting commercial tape recorders and other equipment factory-calibrated such that 0VU = +4dBu. The BBC's standard studio Operating Level was 0dBu, and in that situation the standard VU meter constantly under-reads and, because of the small and compressed range of the meter, the needles barely move at all. So it's no wonder that confused BBC operators decided that the VU was "virtually useless" compared with the wonderful in-house BBC PPM (scaled, somewhat mysteriously, from 1 to 7).

The real problem, then, was the BBC's inept alignment regime rather than the VU meter itself. When aligned correctly to the BBC's 0dBu Operating Level, the VU meter instantly provides useful, meaningful indications of perceived volume, just as intended. So if you come across anyone who tells you the VU is "virtually useless," perhaps give them a Paddington Bear-style 'hard stare' and tell them not to be so silly!

VUs In Modern Productions

That's enough of ancient history and embarrassing misunderstandings — what you really need to know is how useful the VU meter might be in a modern computer-based studio. Today, we're spoiled for choice, with many different meter types available that can focus the user's attention on different aspects of that signal. For example, broadcast PPMs focus mostly on (quasi) peaks. True Peak meters focus on the absolute level of transient peaks when a discrete digital signal is reconstructed as a continuous waveform. The BS.1770 Integrated Loudness meter indicates the perceived loudness over an entire song or programme… and so on. (Speaking of which, it's no accident that the Momentary option in the BS.1770 Loudness Metering suite has a very similar specification to that of the original SVI.)

A VU meter is far simpler than all of those. It's a basic form of RMS meter that conveys a rough and ready impression of the instantaneous volume measured, effectively, over the last third of a second. Also, it really only works properly if the average signal level is close to 0VU. But while that might sound like a disadvantage, in practice it can be very beneficial because the numbers on the scale are largely irrelevant. Really, it's the angle of the needle that conveys the important signal level information and this is translated subconsciously and instantly, so interpreting it doesn't use lots of mental processing (unlike bar-graph meters or numerical readouts). Consequently, mental attention isn't diverted from concentrating consistently on the sound in order to process the eyes — and that's what this business is all about!

If the signal falls too low, the needle drops quickly to the left, and if it's too hot the needle will soon be 'pinned' to the right. In between, it's only really the change from black to red at 0VU that matters. For clean signals, vertical is generally about right, 30 degrees over to the right ("into the red") is too hot, and 30 degrees to the left is too cold. Of course, it's common today to 'drive' equipment into distortion for effect, but if you start by getting the signal to the »

» normal operating levels and add gain from there, you'll generally find it easier to hit that distortion sweet spot. Of course, if you tend to use a piece of gear only for such effects, the 'proper' way to do this would be to calibrate the VU meter so that 0VU is aligned with that sweet spot!

Okay, so the VU meter might not be a one-stop solution for every query concerning signal levels, but it does provide a very good instant impression of the average programme level/volume and everyone knows instinctively when the meter needle is bouncing around in the right area. I've never had to explain to anyone how to interpret a VU meter because it's inherently so obvious! The meter's simple scale and relatively fast reaction time also mean that calibrating the input and output levels within a system, whether a digital, analogue or hybrid one, is unbelievably fast. Source a steady tone and adjust the level to read 0VU, with remarkable precision and zero ambiguity. It's much easier than with a bar-graph meter, for example.

Meter Calibration

As you'll have gathered by now, the many benefits of the VU really only apply if it is calibrated accurately to the system's Operating Level or desired mixing level, and this means that some form of calibrated input level adjustment facility is an absolute necessity. Thankfully, most VU meter plug-ins have a way of adjusting the (digital) sensitivity over a range of, say, -12 to -20 dBFS but, sadly, most hardware VU meters — and especially the cheaper ones — do not. Nevertheless, my preference is for hardware meters where possible, because they don't take up valuable screen space, and if you wire them to a monitor controller's record output then one meter can be used for all your audio sources, even when the computer is switched off or doing something else. (Of course, big VU meters can also look very cool in the studio!)

The challenge is to find a hardware VU meter with a calibrated adjustable input attenuator, as well as having the correct VU meter ballistics — the needle rise and fall times are critical to the accuracy and usefulness of the display, and just putting a VU scale on any old moving-coil meter won't do for mixing applications! Having scoured the audio manufacturers' product catalogues, I'm currently only aware of two hardware units that meet these requirements in both respects. Crookwood Audio Engineering (https://crookwood.com/vu-meters/2u-stereo-vu-meter) offer three VU meter variations using excellent SIFAM movements, all with a switchable attenuator covering a 15dB range. Alternatively, Japanese manufacturers Hayakumo (https://en.hayakumo.tokyo) offer their Foreno VU meters fitted with NHK-approved Fuso meter movements, along with a switchable 15dB attenuation range. (There may be other meters out there that meet the requirements, but if so I've not yet found them.)

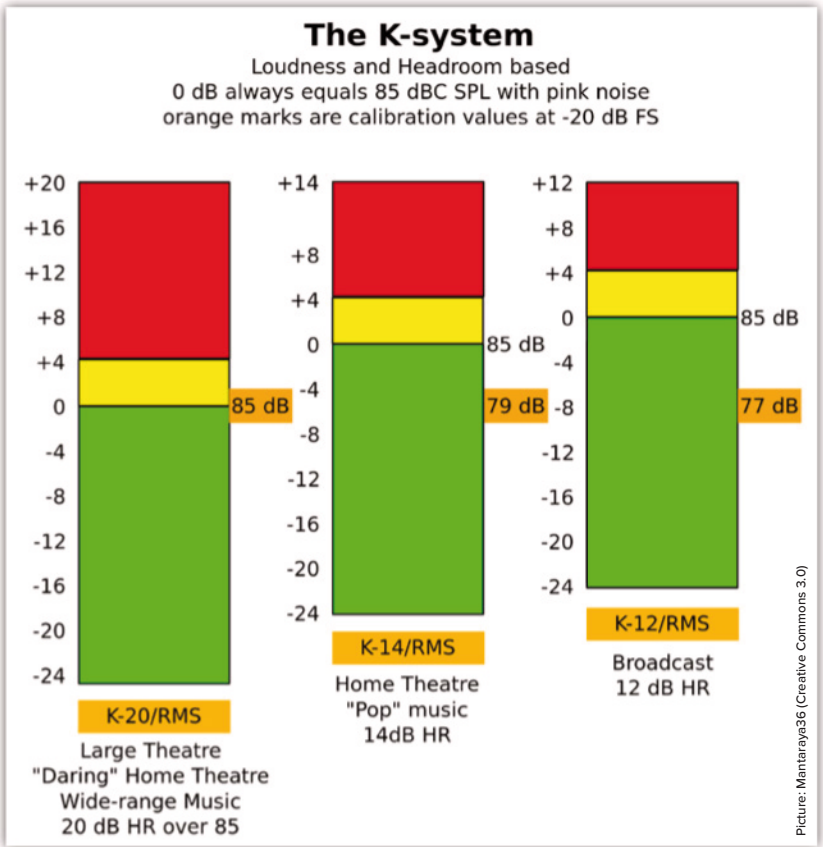
Using VU Meters When Mixing

If tracking through an analogue console, the factory-set alignment of 0VU = +4dBu is a good option that optimises headroom and signal-to-noise performance. If working entirely in-the-box in a DAW, we might choose to track with a nominal operating level of -20dBFS (the American SMPTE standard) or -18dBFS (the European EBU standard), and it's easy to select the desired calibration in most VU meter plug-ins, thus keeping signal levels in the right ball-park. When mixing or mastering, though, it's often desirable to work with a higher overall mix reference level, and this is really where the adjustable meter calibration becomes critical.

For example, if mixing for a streaming service the desired Loudness might

K-System Metering

In 1999, mastering engineer Bob Katz conceived the K-System Meter, in effect a digital replacement for the SVI. I won't go into all its design goals and benefits here, but two key aspects are directly relevant to this article. First, it establishes a defined operating level in the digital environment. Second, there are three alignment options to cater for alternative desired mix levels. On the down side, it usually employs a bar-graph rather than an angled pointer (as explained in the main text, I find the latter easier to use). The K-20 option is equivalent to a standard analogue VU, with 0VU at +4dBu and 20dB of headroom, and is intended for tracking wide-range music. K-14 and K-12 move the mix reference level higher, with correspondingly reduced headroom and dynamic range, much the same as mixing for -14 or -12 LUFS for streaming.



Mastering engineer Bob Katz created the K-System, which is, broadly speaking, a digital equivalent of the VU meter with several refinements, not least catering for three different alignment levels.

Quasi-PPM

Despite their name, none of the European peak-programme meters (PPMs) developed in the 1930s measured what we'd now call 'true peak' levels, so they're more accurately know as 'quasi-peak meters' (QPPMs). During their design it was found that very loud but brief transients were of little technical risk to the analogue transmitters or tape recorders, and that (analogue) transient distortion lasting less than a millisecond or two is inaudible (the generated harmonic overtones are beyond human hearing). Also, registering such brief transients on the meter generally resulted in the broadcast operators setting average levels unacceptably low, reducing transmission ranges and degrading the signal-to-noise ratio. So a little under-reading of transient levels was (and in analogue systems still is) seen as a positive benefit rather than a negative, and none of these PPMs respond to the very fastest transients; most dramatically under-read peaks shorter than 10ms or so. The same is not true in digital systems, because even the briefest transient overload results in aliasing artefacts that have an unnatural harmonic content and so are easily detected by the listener. This is why True Peak (TP) metering is now mandatory for services adhering to the Loudness Normalisation standards, and true peaks may not be higher than -1dBTP, thus guaranteeing no transient overloads whatsoever.

The PPM — or rather QPPM! —is a meter design that pre-dates the SVI, but QPPM meters are still widely used today. They're arguably more accurate than VUs for gauging signal levels in broadcast, but they were far more expensive in the 1930s and '40s, and their behaviour and scale don't bear the same resemblance to human perceptions of loudness.

be -14LUFS, and you could dial that into a loudness meter and work just with that. However, the constantly varying LUFS numbers aren't as easy or fast to interpret as a simple VU meter, and I find that simply by adjusting the VU meter's sensitivity by the appropriate amount, I can tell instantly when I'm mixing in the right ball-park for that target loudness. It's also much easier to check if the chorus is sufficiently louder than the verse, if the snare drum is balanced to everything else, and whether the vocal needs to be pushed up or down a smidgen — just from the experience of looking at different material on VU meters.

So, the pertinent question is: what's the right VU alignment for mixing? Obviously, it depends on what your target level is but a simple approach is to run 30 seconds of pink noise through the mixer (whether a physical one or your DAW) with both a VU meter and an Integrated Loudness meter registering the output level. Adjust the level of pink noise through the mixer until the Integrated Loudness meter indicates the desired target level (-14LUFS, say). With the pink-noise now at the target level, adjust the VU meter's alignment to place the needle close to the -2VU mark (ie. vertical).

Once calibrated in this way, it should be relatively easy to keep the mix levels very near the desired target level just by glancing at the VU meter. Naturally, this will take a little practice and experience, and you may need to fine-tune the exact alignment for your particular mixing techniques. But this solution should work well and most people will find it much easier and less stressful than focusing on a loudness meter.

So, despite being 83 years old, the VU isn't 'virtually useless' at all: it's still a very practical, highly appealing, and valuable tool for every audio mix engineer.

WAVE ARTS
DEFINITIVE SOUND

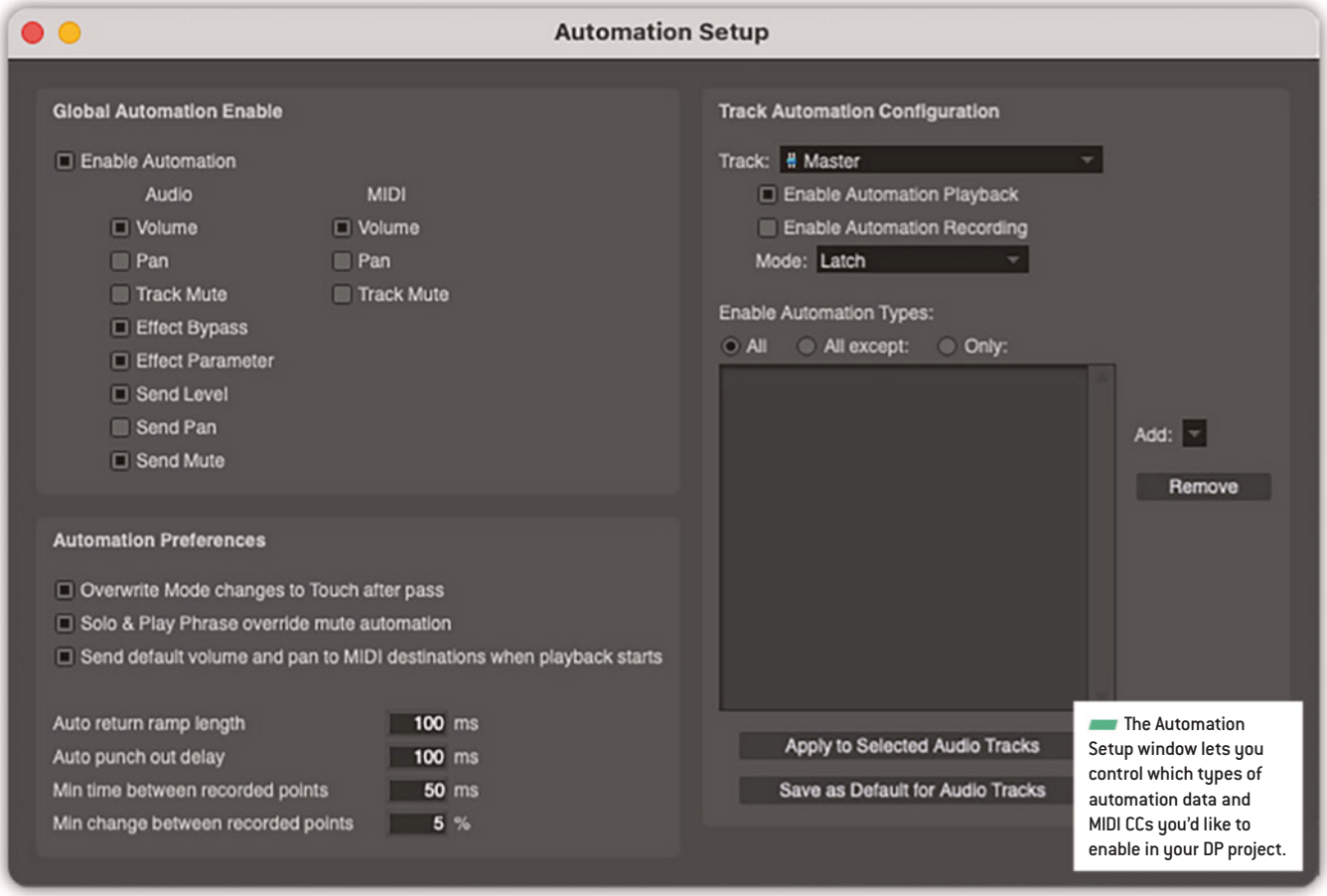
PANORAMA 7
virtual acoustics processor

MULTIDYNAMICS 7
powerful multi-band dynamics processor

FINALPLUG 7
lookahead peak limiter with loudness metering

wavearts.com

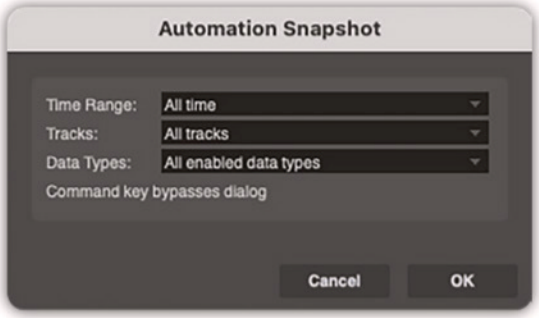
DP's Snapshots allow complex automation to be generated at a stroke.



The Automation Setup window lets you control which types of automation data and MIDI CCs you'd like to enable in your DP project.

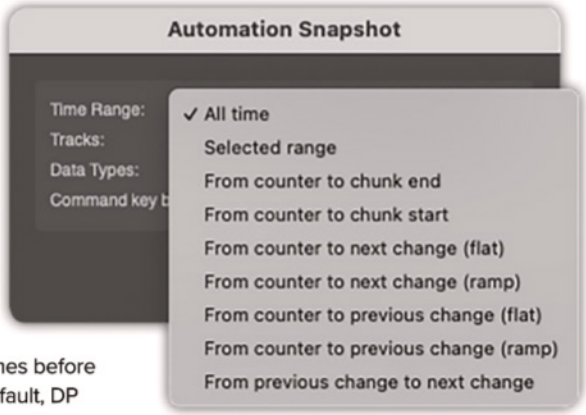
MATT LAPOINT

Digital Performer provides multiple ways to create automation data. Fader and plug-in parameters can be captured with mouse moves, control surfaces, and knobs and sliders on MIDI controllers (programmed with DP's MIDI Learn feature). The pencil tool is also a precise way to insert automation and MIDI CCs in DP, with its control points, lines and preset curves. But there's another fast and powerful way to insert timeline automation: Snapshots.



You'll see this Automation Snapshot window when taking a Snapshot.

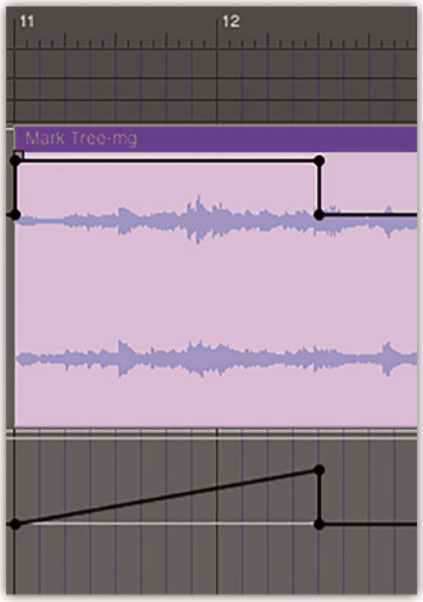
Much like a camera, a Snapshot captures and inserts automation anchor points in the timeline. The resulting automation points recall parameter states at the time the Snapshot is taken. Snapshots are not only important for recalling exact automation levels; the anchor points can also be used as pivot points for additional automation that comes before or after captured points. By default, DP will chase all automation points, but any parameter type can be excluded from event chasing, if desired. Use Setup / Event Chasing for MIDI controllers and Setup / Automation Setup for audio parameters.



The Time Range menu gives you plenty of options for how to apply the Snapshot.

Data Mining

When working with Snapshots, it's important to differentiate between audio automation data and MIDI Continuous Controllers (CCs). Audio volume, pan, effect and virtual instrument parameters are all part of the audio automation system. MIDI volume (CC7), pan (CC10) and track mute messages are the only MIDI parameters captured with Snapshots. CCs like modulation (CC1) and expression (CC11) are omitted from Snapshots. These parameters are captured by recording, overdubbing, drawing and reshaping MIDI data.



Flat versus ramp Snapshot automation.

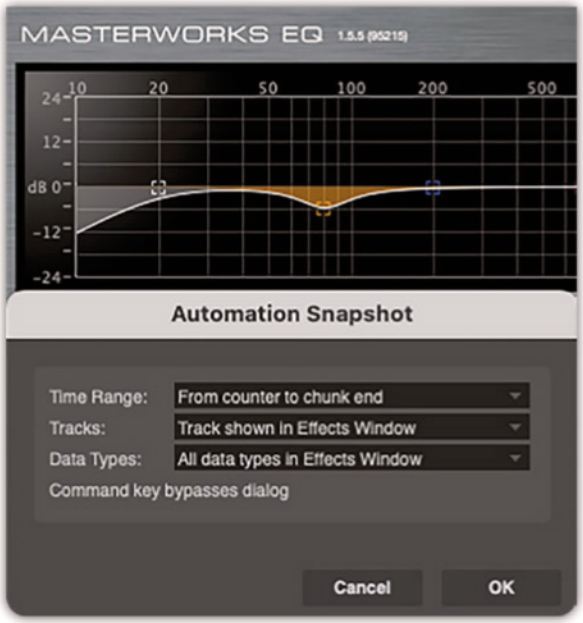
Found in the Setup menu, The Automation Setup window displays the various MIDI and audio automation data types that can be enabled or disabled globally in a project.

It's important to know how this window relates to Snapshots. Enabled data types captured by Snapshots will play back and chase the wiper position. Make sure the automation play button is enabled in the mixer, editors and effects windows. If the play button is disabled, or the parameter is deselected in Automation Setup, the lines representing the automation will appear with dashes and not as dark lines. In Automation Setup, non-enabled data types will be ignored when a Snapshot is taken. If you accidentally leave out a data type when taking a Snapshot, you can simply re-enable that data type and take another Snapshot with that data type specifically targeted in the Snapshot window (more on this later).

Snappy Snaps

Snapshots are taken with the camera icon button at the bottom of DP edit windows. The default keyboard shortcut is Ctrl+' (Mac) or Windows+' (PC). (The quotation key is just to the left of the return key.) Adding the Cmd key (or Ctrl key for Windows) while invoking this shortcut bypasses the Snapshot dialogue and uses the previous settings in the window. The Mixer, Sequence Editor and MIDI Editor are the primary windows for taking Snapshots, although they can also be taken from the Tracks window, Event List, QuickScribe, and nearly everywhere in the program.

Important tip: the Snapshot window targets the currently active (focused) window, which will dictate and potentially limit what tracks, data types and selections will be captured. Snapshots in the Mixer, Sequence Editor and MIDI Editor will provide the most control, as discussed below.



Open a plug-in effect or VI window to take a Snapshot of its parameters. In this example, the MasterWorks EQ is being targeted.

Snapshots are applied over a time range. The Time Range menu provides useful options to choose from: 'All time' is the full length of your project, while 'Selected range' is based on your highlighted time range selection. The two 'From counter' options use the wiper position (counter) to define the start and end range. Chunk start and end boundaries are determined by the first and last data locations in your timeline. The Chunks window (Shift+C) provides editing of chunk start and end boundaries. The next four Time Range menu options provide counter (wiper) anchor points, with options to produce flat or ramp results. The flat options produce a static value, whereas ramp options produce a linear change. The last menu option uses existing data points to determine the range.

The Tracks menu gives you control over where the Snapshot automation will be inserted.

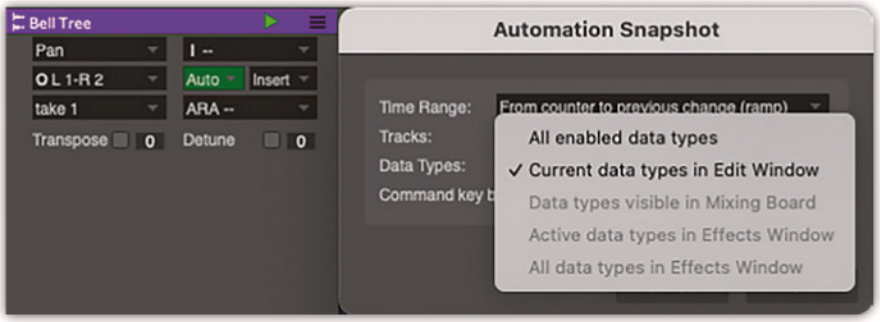
Target Acquired

The Tracks menu in the Snapshot window has options for choosing which tracks are targeted during a Snapshot. 'All tracks' and 'Selected tracks' are self-explanatory. The availability of the remaining menu choices will be based on the focused DP window when taking the Snapshot. For 'Tracks shown in Edit Window', take the Snapshot from the Sequence Editor or MIDI Editor. Use the track selector in DP's edit windows to show/hide specific tracks for targeting with Snapshots. For the Mixing Board option, focus on the Mixing Board by clicking on its title bar. To Snapshot parameters for a specific plug-in effect or virtual instrument, open its effect window. With this Data Type setting, all MOTU and third-party plug-in parameters are captured during a Snapshot. Published parameters can be viewed and edited in the Sequence Editor in each track's Edit Layer menu.

Data Types

The Data Types menu provides options for choosing which types of data are targeted during a Snapshot. The 'All enabled data types' option includes all parameters enabled in Setup / Automation Setup. 'Current data types in Edit Windows' is based on the parameter shown in the active edit layer. Like the other Automation Snapshot menus, the availability of the remaining menu choices is based on the window that is in focus when the Snapshot is taken. Active data types are types that are already present in the track. In the Sequence Editor, use 'All data types' when taking a volume Snapshot for a MIDI track and audio track at the same

»



The Data Types menu lets you choose the types of data that will be captured by the Snapshot. In this example, only Pan automation will be captured for the Bell Tree track because it is the current (visible) data type.

» time. Even if the MIDI track is viewing the notes layer, the volume Snapshot will still be successfully captured when you click the camera icon.

Instrument & Mixer Snapshots

DP11 introduced instrument tracks, which combine both MIDI and audio in a single track. When taking volume and pan Snapshots of instrument tracks in the MIDI Editor, DP uses audio automation rather than MIDI CC7 (volume) and CC10 (pan) because audio automation provides higher precision than MIDI data. When using Snapshots with instrument tracks that contain data on multiple MIDI channels, only the overall volume, pan and track mute states are captured for the instrument track.

To capture individual MIDI channel automation, select all of the MIDI data in the instrument track and use the command Split by Channel. This will separate the MIDI data to individual MIDI tracks, each assigned to the corresponding instrument. Conversely, separate MIDI tracks assigned to the same instrument can be selected and the command Merge by Channel will absorb the MIDI tracks into the single combined instrument track. The MIDI data will stay assigned to the original MIDI channels.

The mixing board is another useful place to take automation Snapshots. All data types and visible data types can be captured. To change the visibility of data types, use the mixing board mini-menu to show or hide parameters as desired.

As mentioned earlier, if a plug-in or virtual instrument window is targeted with the Snapshot window, all of the parameters will be captured. In the Sequence Editor, use the track's Edit Layer menu to see a list of the published parameters captured from the effect Snapshot. Select specific parameters to create new automation data or edit it. In the Sequence Editor, track lanes can be used to view multiple automation types at the same time.

Parameter Overload

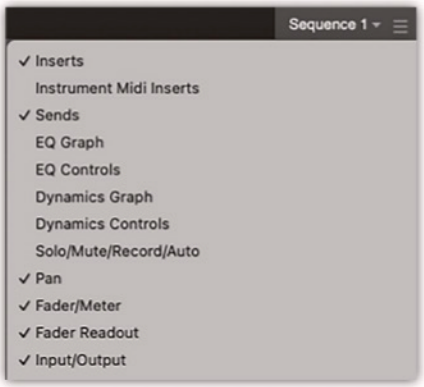
Using Snapshots for spot effects in plug-ins or VIs with an extremely high number of parameters can get cumbersome. In cases where you only need to capture a select number of parameters, use MIDI Learn to assign these. To then Snapshot these specific parameters, use 'Active data types' in the Snapshot window's Data Types menu. Also, note that while algorithmic reverb plug-ins like eVerb and Plate lend themselves well to spot effects, allowing you to automate the wet/dry levels and the length of reverb tails, the ProVerb plug-in from MOTU can't automate the impulse response length because of the offline audio rendering that is required when the sample waveform is altered.

Mix Mode

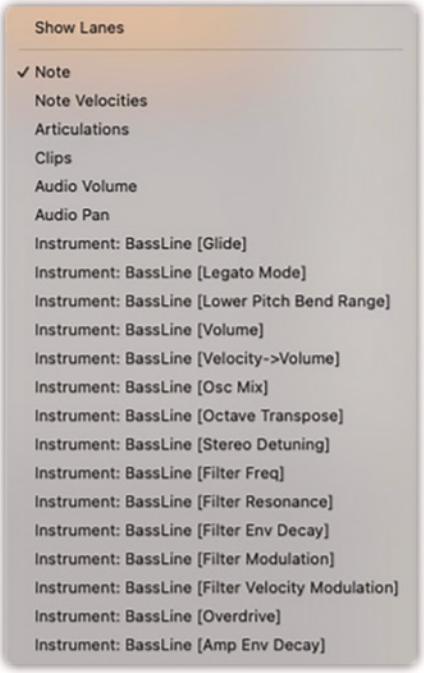
Digital Performer has a powerful feature in the mixing board called Mix Mode. By default, this mode is turned off. Choosing New Mix will keep the tracks, routing, and sends but remove automation data and insert plug-ins. Each mix can have completely different automation states captured by the Snapshots window. Duplicate Mix could be used to create an alternate mix with different captured Snapshot settings. It's a powerful way to quickly create different mixes while staying in the same project.

In Summary

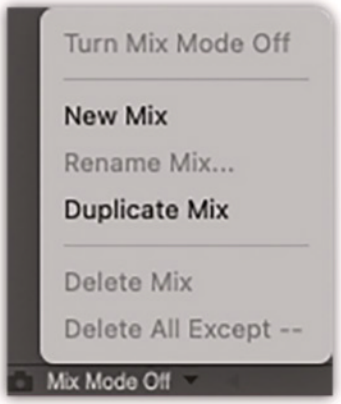
The Snapshot window is a fast and powerful way to control automation data, making possible everything from minor tweaks in an individual track to project-wide automation changes across all tracks in the DP timeline.



Use the Mixing Board mini-menu to show and hide sections as desired, when using the 'Visible data types' Snapshot option. In this example, track mute has been hidden, so it will be excluded from the Snapshot.



A track's Edit Layer menu in the Sequence Editor shows a list of published parameters captured in the track by an effect or VI Snapshot.



DP's powerful Mix Mode menu is found in the Mixing Board, to the left of the horizontal scroll bar.

Decapitator

The Secret Weapon of Top Mix Engineers

From subtle to extreme, analog saturation is an integral part of great mixes. That's why we studied all the analog classics, and painstakingly created a plug-in that brings the best of analog saturation to your studio. Choose from five different models to add analog character to every track and bus in your mix.

"Decapitator is instavibe-in-a-box."

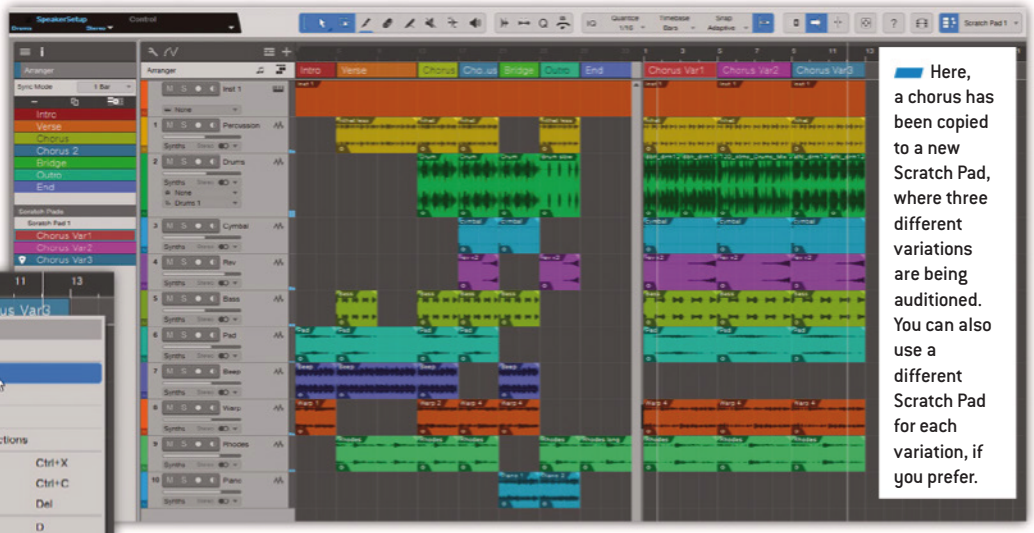
– Fabrice "Fab" Dupont (André 3000, Shakira, Jennifer Lopez, Marc Ronson, Isaac Hayes)

---

soundtoys

Free 30-Day Trial
at www.soundtoys.com

The Scratch Pad is your ticket to fast, easy, non-destructive experimentation!



Here, a chorus has been copied to a new Scratch Pad, where three different variations are being auditioned. You can also use a different Scratch Pad for each variation, if you prefer.

ROBIN VINCENT

PreSonus introduced Scratch Pads to Studio One version 3, about eight years ago. They are an innovative form of version control, allowing you to experiment with different directions within a single project. In many other DAWs, you would have to scatter the timeline with little experiments as you rework a chorus, or your project folder would overflow from a stream of saved revisions with increasingly cryptic or desperate filenames. Scratch Pads let you experiment with your arrangement without losing sight of your work's main thrust. They are also really easy to miss.

In this workshop, we're going to get into the Scratch Pads. We'll look at how they work and, perhaps more importantly, how you can use them to develop your music.

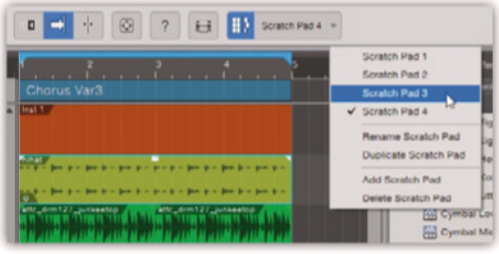
Scratch That

The biggest reason Scratch Pads exist is because we are messy and destructive beings and cannot be trusted to take good care of our compositions. As we develop our music and start working with ideas, we might find ourselves wanting to focus in on a section for experimentation. We don't want to destroy what we've already created and so, normally, we have two options. We can copy the section to another part of the timeline, make our changes and copy it back in. That can work well, but it can also get complicated, and you need to be very tidy to make sure it fits back in place, or you might have to do a lot of clip shuffling to get it right. The other option is to save the project as another file and work on it safely in the knowledge that the original is not being tampered with. This works OK, too, but listening and comparing between two projects is messy and time-consuming, and recombining them again is difficult. Both these options require you to have almost inhuman project management skills and a level of togetherness that I rarely see in a studio.

So, into this mess enters the Scratch Pad, and you wonder why no one thought of it before and why no other DAW has adopted it. Maybe everyone else has got their shit together.

Padding Out

A Scratch Pad is an alternative arrangement page within Studio One's Arrange view. It splits off to the right of the main Arrange View and is in the same format, following the same tracks and using the same editing tools. You can throw the playhead into the Scratch Pad by clicking in its timeline ruler. Dragging the dividing line reveals more or less of the Scratch Pad, although you never fully lose the main arrangement underneath. Scratch Pads have no impact on the mixer console, as the tracks flow through from the main arrangement to the Scratch Pads, so you should see it not as an alternative project but as an alternative arrangement.



Scratch Pads are accessed from this button in the toolbar. You can have as many as you want, and can duplicate existing ones to try variations on your variations.

The main concept is that it's a space where you can drag audio and MIDI clips, sections or elements in order to experiment with them safely away from your main arrangement. When you drag in clips, they are automatically copied without you having to hold the Alt/Option key, so you are never in danger of losing the original. However, you can also drag in whole sections using the Arranger track, which handles the Alt key differently. If you drag without the Alt key held, it copies, but if you hold the Alt/Option key, it cuts, removing the material from the original arrangement. Weirdly, this is the exact opposite of what happens when you move things around in the main Arrange page.

It should be noted that the Arranger track combines perfectly with the Scratch Pad, making the movement and transference of sections between main and Scratch Pad arrangements seamless.

You can treat the Scratch Pad just like the standard Arrange page. All the slicing and dicing tools are there; you can pull in other loops, record audio, enter MIDI notes, and access the piano roll, audio editors and mixer. However, this does not extend to the Score Editor. In practice, it works exactly as if it was your main Arrange page. You can have as many Scratch Pads as you like, adding new ones or duplicating existing ones from the button on the right of the toolbar. You can only view one at a time, and you can rename them to keep track of your flights of fancy.

Radio Edit

If you have a finished song with a neatly annotated Arranger track, you can use

a Scratch Pad to try out some radio edits or shortened trailer or jingle versions of your track. Studio One has some workflow shortcuts to make this really easy. Rather than creating a new Scratch Pad and dragging sections to it, you can drag to select all the Arranger sections, right-click and select 'Copy (or Move) to new Scratch Pad'. This gives you an exact duplicate of your song that is ready for some alternative edits.

The Arranger List in the Inspector window gets populated with the sections from all your Scratch Pads underneath the main one. So, you can whizz around the Scratch Pad using the Arranger List, just like you can in the main window, making it really easy to navigate. You can enable Sync Mode to keep the movements between sections quantised to a bar or two. Then you can make edits in the Arranger track, slicing verses in half or reducing the length of the chorus, and using the right-click menu option Delete Range to remove unwanted sections and have everything else move in to take up the slack. All of this without having to slice up the individual clips or change the arrangement. Working with the Arranger List and then applying those changes to the Arranger track in the Scratch Pad is a really fast way to cut down a song into bite-size pieces while retaining the gist of what you're doing.

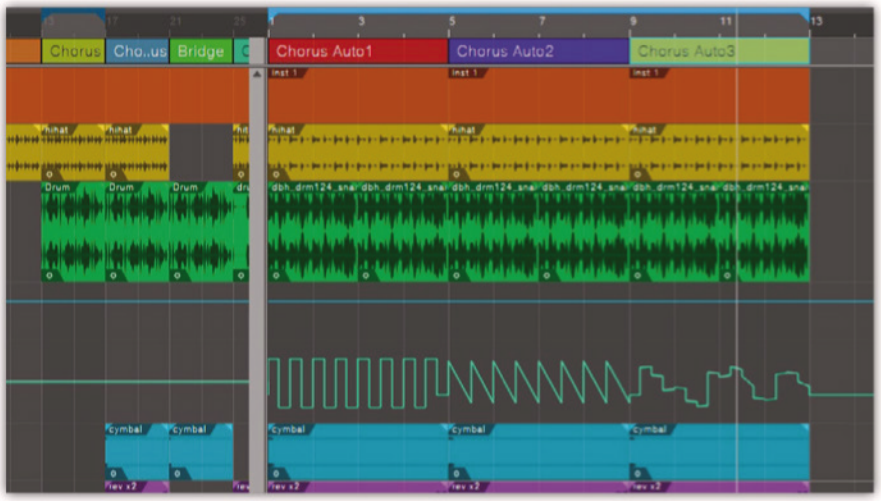
Loop Variations

Another great use is in the auditioning of variations. Say you've got a chorus with a particular drum loop, and you'd like to try out some other options. You can copy the chorus to a Scratch Pad, remove the loop and replace it with other loops from your library. As you are just dealing with the chorus section and not the whole song, you can duplicate the clips out a few times and add different drum loops into each duplication.

This is a fine and efficient way to handle the auditioning, which doesn't fill up your main timeline with a scattering of multiple experiments; the Scratch Pad is awesome for this. However, if you are using the Arranger track to pull the chorus across, then you don't really want to be changing its duration, or you might have some difficulty pulling it back into the main arrangement. So, perhaps a more efficient way is to audition different loops or versions across multiple Scratch Pads.

Once you find a loop you like, you can copy the new chorus to another Scratch Pad and try out another loop or two. Once you've got a few Scratch Pads working the same chorus with different loops, you can swap between them while Studio One is playing back by selecting the pads from the menu. It's a much better way of auditioning variations and changes and homing in on the one that's going to work the best.

Having said all that, Studio One is famous for having three different ways of doing something when one would be plenty. If we go back to using a single Scratch Pad, we can make resizing the chorus less problematic by using the Arranger track to section out the duplications. That way, the chorus stays the right length, and you can simply choose the one you want to drag back into the main arranger. You have the added bonus of being able to use the Arranger track Inspector to quickly leap from section to section to audition those loops, or simply double-clicking in the Arranger track.



As well as housing audio and MIDI clips, Scratch Pads can also contain automation, allowing you to experiment with different automation passes or modulation treatments.

Whichever way you find works best for you, this is infinitely better than messing up your original timeline with a scattering of clips and samples.

Listening

But wait, there's another approach that could be even more efficient, depending on your point of view. In the main toolbar is a speaker icon that you may know as the Listen tool. When you use it in the main arrangement window, you'll find that it will solo whatever clip you click it on. In the Scratch Pad it has a very different and much more interesting function. If you set your main project playing back, so the timeline in the Scratch Pad is greyed out, you can use the Listen tool to teleport clips from the Scratch Pad into the main arrangement. It only happens while the mouse button is being held down, and you can see a grey version of the clip overlaid on the track. Once you release, the track is returned to normal.

So, you can loop the playback around your original chorus and simply click on the new drum loops you're auditioning and have them play in the right place within the original track. It doesn't solo the clip like it does when you are working in the active arrangement; it plays the clicked-on clip in place. This also works going the other way, to teleport clips from the main arrangement into the Scratch Pad.

Automation

Scratch Pads don't just deal with clips, they can also deal with automation. If you're anything like me, then you'll find that you can easily mess up a track by experimenting with automation. The Scratch Pad is perfect for trying out and comparing how something would sound if it was modulated. You can check out different ways of moving parameters, you can record different passes and tweak them differently, and you can compare what the effect is in the context of the song. It can be like building up a library of different captured parameter performances that you can then drop into your song whenever and wherever you like. However, the Listen tool doesn't work for automation, which is a shame.

Hopefully, I've been able to point out that while the Scratch Pad does the job of keeping your project tidy, it offers multiple ways of working on your music without losing the focus of your song. Perfect for experimentation but also great for keeping your shit together.

Pro Tools has upped its MIDI game. What can the new MIDI plug-ins do?

JULIAN RODGERS

It is widely held that MIDI isn't something that Pro Tools does best. Looking at the history of the program and its counterparts like Logic and Cubase, this is understandable. Cubase and Logic started life as MIDI sequencers, which over time gained audio capabilities as computers became fast enough to handle multitrack digital audio processing natively. Pro Tools has a rather different history, in that it started life as a DSP-powered audio workstation system at a time when computers weren't powerful enough to perform these tasks without help.

Over time, Pro Tools gained MIDI functionality, and ultimately, MIDI sequencing software and audio workstations converged into the modern DAWs we recognise today. Much of the negativity some people attach to the MIDI side of Pro Tools is based on received wisdom about MIDI as it was in Pro Tools years ago. When I started using Pro Tools 5 MIDI was, to put it politely, basic; but that was a long time ago, and things have changed a great deal. While some MIDI features are still missing from Pro Tools, one of the biggest gaps was addressed in Pro Tools 2024.3 with the addition of MIDI plug-ins, along with an overhaul of MIDI routing to accommodate them.

Plug-in Power

There are now six MIDI plug-ins included in Pro Tools. These plug-ins share the same AAX architecture as audio plug-ins in Pro Tools, and are accessed from the insert slots on Instrument tracks. They cannot be instantiated on MIDI tracks because MIDI tracks lack the insert slots in which to instantiate them.

The fact that the MIDI plug-ins co-exist with AAX plug-ins and virtual instruments might seem counterintuitive: MIDI and audio I/O are kept distinct from each other in Pro Tools. But this is a distinction which isn't made quite so rigorously in other DAWs. For example, in Reaper a track is just a track, with no need to designate it as audio or MIDI. I like having MIDI plug-ins in the insert slots, but it's important to note that a MIDI plug-in following an instrument plug-in will interrupt the signal flow and result in silence — plug-in order is important.

Three of the newcomers are 'in house' MIDI plug-ins, and these are included with all tiers of Pro Tools, including Intro. These Avid plug-ins are intended to perform utility processing, and at present comprise Note Stack, Pitch Control and Velocity Control. At first sight it might seem that these plug-ins are duplicating facilities already available in Pro Tools, so how are they different from what is already on offer?

The original way to manipulate MIDI data, other than by manual editing, was by using the Event Operations window. This window received an interface overhaul in Pro Tools 2023.6; rather than each page being selected from a drop-down menu, they are now accessible simultaneously, with each section accessed using disclosure triangles, which is much more convenient. As well as control of quantise, this window offers velocity and pitch manipulation, and as such, does overlap with what is offered by the new Avid MIDI plug-ins. We can see Event Operations as the MIDI equivalent of AudioSuite processing, in that the results are non-real-time and are 'baked into' the MIDI clip.

Property Development

An alternative to directly manipulating MIDI data on the timeline is to use MIDI Real Time Properties. As the name suggests, this allows real-time processing of MIDI data on a track or clip basis. The processing available is utilitarian in nature: quantise, transpose, note delay/advance, duration and velocity control. The Real Time MIDI Properties feature is much more usable if you enable the 'Display events as modified by real-time properties' preference, which means that while the underlying MIDI data can remain unmodified, you will actually see what you hear.

With two very capable ways of manipulating pitch and velocity already available, what advantages are there to accessing similar features using the new AAX plug-ins? Apart from the convenience of putting MIDI processing in the same place as audio processing — in the plug-in insert slots — the biggest immediate benefit is that everything in a MIDI plug-in is automatable. This

The new Note Stack and Pitch Control MIDI plug-ins are included in all versions of Pro Tools.

MIDI Real Time Properties also allows you to adjust MIDI data as it's playing, but with less flexibility than the new plug-ins.

is well illustrated by Note Stack. This plug-in seems rather dry in nature: on first inspection, it allows you to stack notes by octave and semitone. The transposition isn't related to key signature, and as such, it seems like it wouldn't bring much to a composition. However, the ability to automate parameters makes things considerably more interesting. For example, the abilit to automate the offset while also automating the Note parameter, which enables/disables the note, encourages experimentation and introduces elements of unpredictability. However, it is the Probability parameter that brings the most immediate rewards. Setting Probability to a value lower than 100 percent means that Note controls the likelihood of that note sounding, and can introduce complexity and interest to a part.

The Pitch Control and Velocity Control plug-ins add greater flexibility compared to Real Time MIDI Properties. The ability to include or exclude MIDI data based on pitch or velocity in both plug-ins opens up interesting possibilities. Pitch Control offers transposition in key with the Key Signature Ruler, and Velocity Control offers more flexibility than the two-parameter Vel section of Real Time MIDI Properties. Whereas the latter simply allows velocity data to be globally added to, subtracted from or scaled, Velocity Control can be much more selective, allowing you to achieve something closer to compression. For example, setting the Velocity Range to only process notes above a velocity of 100, and setting Scaling to 80 percent, means the dynamics of lower velocities are left unaffected while notes with velocities greater than 100 (the 'threshold') will be softened.

In the screenshot, I have Velocity Control set up to as a MIDI compressor, acting on the Inst 1 track. In the I/O section of the lower MIDI track, I'm recording the output of Velocity Control using the MIDI Chain. This is central to how this new system works. It's similar to an internal bus for MIDI, and as well as allowing MIDI data to flow through a chain of MIDI plug-ins, it can be used to route MIDI data between tracks, for example to record the MIDI output of an arpeggiator.

Party Time

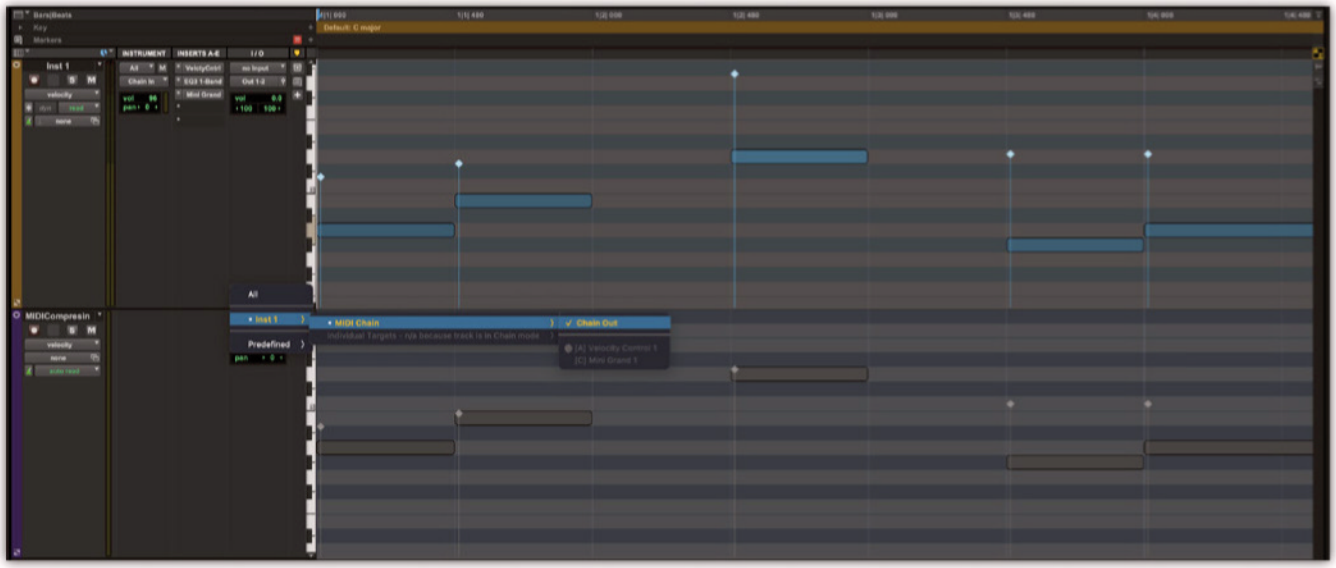
This brings us neatly to discussing the third-party MIDI plug-ins which have been introduced so far. The fact that these are AAX plug-ins, and that half of them are from third parties, indicates that the future will include more offerings from outside the Avid stable. These third-party processors are focused more on generative MIDI processing, and include Modalics EON Arp, AudioModern's Riffer and Pitch Innovations' Grooveshaper Lite. Rather than processing your MIDI, these are designed to generate new MIDI data of their own, and since you you can print their MIDI output to a track via the MIDI Chain and then edit it, it's very much a case of keeping what you like and changing what you don't.

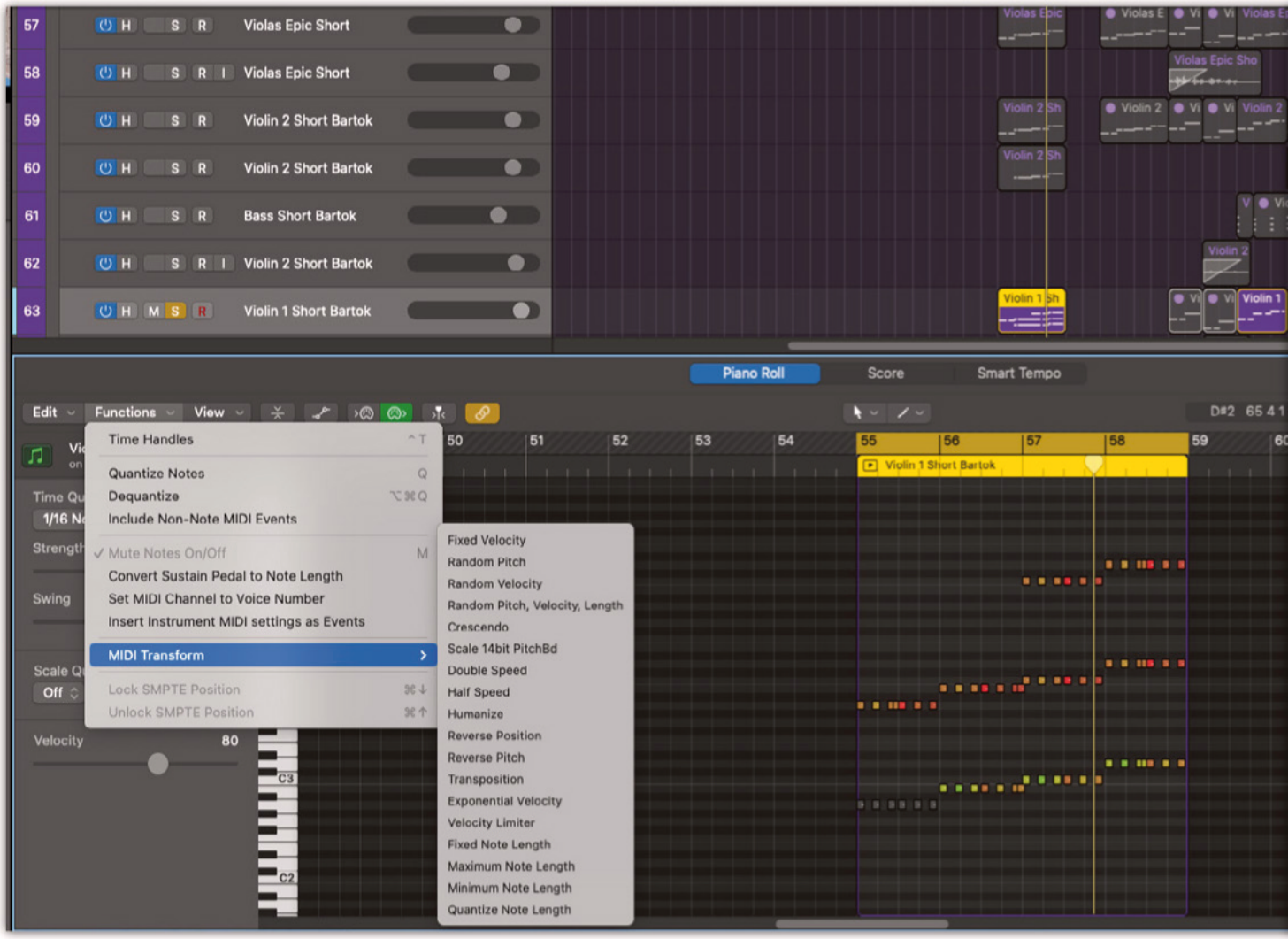
The Modalics arpeggiator is particularly welcome; it's deep, and when used in combination with Note Stack, things can get really interesting. In line with the 'idea generation' role of these generative plug-ins, Riffer and Grooveshaper Lite both incorporate a dice button which will generate a random starting point from which to develop ideas. In the future I'd expect to see more products from third-party developers ported to AAX. Given Avid's investment into the producer market, particularly with the introduction of Pro Tools Sketch, MIDI production is an important area, and these new MIDI options open up lots of scope for new features and products.

One thing I would like to see, though, is a MIDI delay plug-in. I've worked around its absence by duplicating a track and using MIDI Real Time Properties' Delay function to create MIDI delays, but this is rather labour-intensive, and a dedicated plug-in might have extra features like individual MIDI Chain outs for each delay tap. And if each tap could have a Probability parameter, some very cool, unpredictable delay effects could be created...

Generative AI has been developing at pace, and MIDI plug-ins seem particularly well suited to this technology. Where audio artefacts are often the limiting factor on AI audio, these constraints don't apply to MIDI, so MIDI plug-ins might have arrived at an ideal time.

Here, Velocity Control is processing the MIDI data on the instrument track (top), and recording the processed MIDI notes onto the track below, via MIDI Chain.





Add a human touch to your MIDI parts with Logic's MIDI Transform.

SAM BOYDELL

Logic Pro seems to acquire amazing new features in every update, and its bundled plug-ins have been thoroughly modernised. However, there is one feature that has remained untouched for over a decade, and that is MIDI Transform. Yes, its user interface is rudimentary at best and yes, the workflow if encapsulates is dated, but if you do a lot of work with MIDI, it can still prove an indispensible feature.

Transformers

The aims of MIDI Transform are twofold. First, it offers a much faster way to edit MIDI data than manually altering each note, and second, it allows us to bring randomisation, or more importantly humanisation, into our often rigid productions.

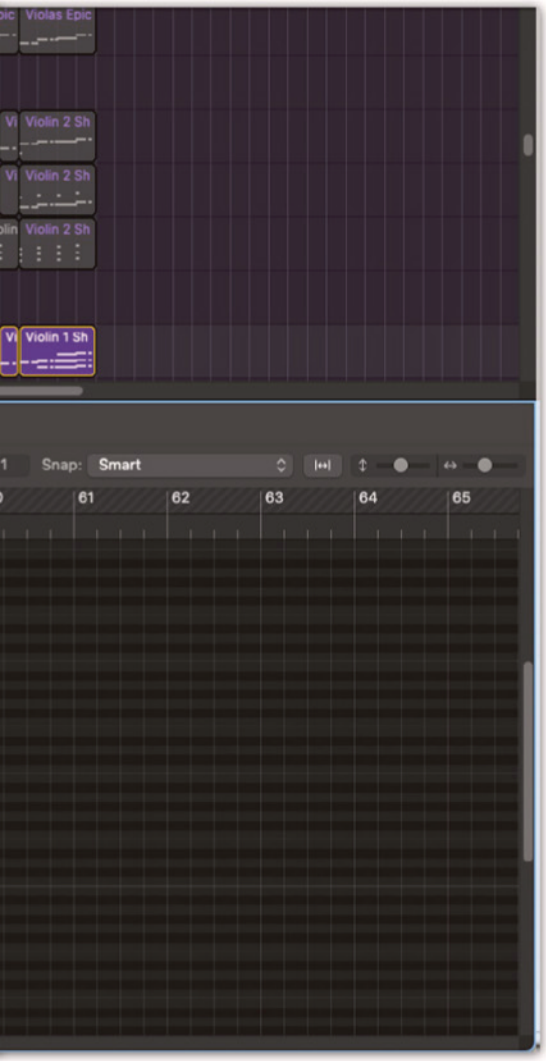
Let's start by taking any MIDI region and double-clicking it so the piano roll pops up. As seen in Screen 1, there is a menu to the left-hand side called Functions that leads to a drop-down list with the item MIDI Transform on it. As you hover over, you can see that there are various things we can do to Transform our MIDI.

Here, we will look into the Humanize function, as shown in Screen 2. This is a particularly exciting one for composers because, as it says on the tin, it introduces randomisation to your MIDI data to bring about a more 'human' feel. This can be crucial to creating parts that sound realistic.

More Than Meets The Eye

Before we dive into the technicalities... Why is it we love musicians? Well, it's their feel, it's their timing, it's the emotion they impart on each moment — and, for better or worse, when we make music inside a computer, this feeling is easily lost. For example, an orchestra of human musicians would never all land on the beat within a millisecond of each other, nor play the notes for exactly the same length and at the same velocity. And we wouldn't want them to, because those differences are what create the depth and dynamics we love. When it comes to computer-based music, we have to break down exactly those qualities of a human performance into MIDI information.

Now, let's now go back to our Humanize function. You will see a series of Events that we can manipulate. As we are in the Humanize preset, only certain parameters are available to us; primarily Position (where the start of the note sits), Velocity (how hard the note is played) and Length (duration of note).



Screen 1: The MIDI Transform menu offers a range of treatments for your MIDI data.

Underneath, you will see what Operations we can do to each of these parameters. Logic has already selected '+-Rand', which will randomise values above and below their current value. Underneath that is a series of numbers that will dictate the range within which we wish to randomise, based on a standard position clock (hh:mm:ss:ms).

In the middle of the page, you will see three blue lines with dots on them. These allow you redirect or swap certain parameters with others. These won't be used in our example, but they show that you can affect almost any parameter using this plug-in.

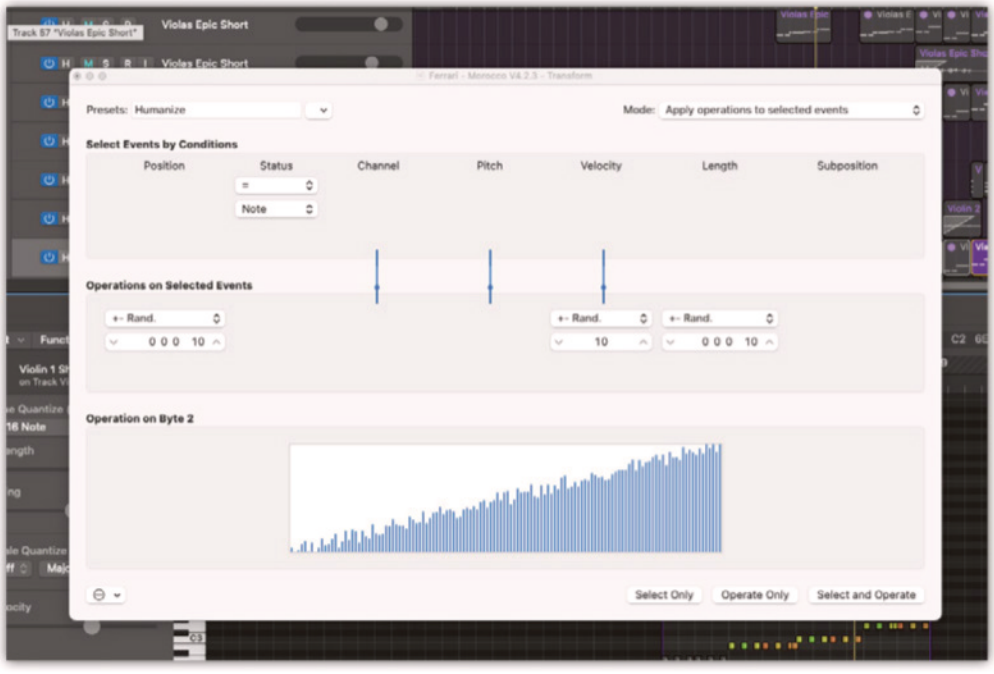
At the bottom of the page we have a visual representation of the operation we are performing, based on the 0-127 range that MIDI allows us, followed by execution buttons reading Select Only, Operate Only and Select and Operate. Select Only simply selects all the MIDI notes that are in view in the piano roll. You can also select from the main environment window with the plug-in open, which makes sense of the other two options: if you wanted to transform only certain notes that you have already selected, you would just choose Operate Only, for example.

Transformers

In Screen 2, from a piece I'm working on, I've selected all the violins that are playing short plucks at the same time, and opened up the Humanize preset. I'm going to modify note positions by keeping the preset on +-Rand, but reducing the range from 10ms to 6ms (10 always feels a little too much to me). Then, I'll put Velocity to zero (we will come back to this later) and Length also to 6ms. Clicking Select and Operate makes the relevant parameters for the visible notes move randomly, between the ranges I set. Important note: we want the first note of every MIDI Region to be in time lest it not play, so we must select all the first notes played and quantise them to the beat.

Next, I'll open the Random Velocity preset. This preset gives you greater control over the range of velocities because you now have two numbers to adjust: an upper and a lower value. The best settings will depend on the material, but for violin libraries (as in my example) anything from 0-60 tends to be very quiet, and as this is a loud part of my track, placing the range at 80-127 is a better starting point.

Screen 2: The Humanize window lets you apply randomisation to your notes' onsets, durations and velocities.



Robots In Disguise

And that's it! You will have to go in and individually adjust some of the changes that have been made to better suit your material, but this should have greatly increased the realism already. I personally do this with almost all my sampled instruments, from drums to synths. As producers we often copy and paste parts, over-quantise performances, use the same sounds and phrases twice or even draw parts in with a mouse. Simply, this is the fastest way to make your parts sound individually played in, which creates that much needed sense of depth and dynamics in a song to keep listeners believing until the end.

The MIDI Transform function is a very useful option with all sorts of roles to play in music creation. Further to the above, I often use the Fixed Note Length preset for those awkward synth parts that sometimes don't trigger as expected if the notes overlap. Presets such as Crescendo also have an obvious application (as long as you humanise them afterward!). Now, it's up to you to go and experiment.

Cubase's VocalChain can polish and add character to your vocals in an instant.

JOHN WALDEN

For Artist and Pro users, the return of the Vocoder plug-in (which we explored in the March 2024 column) was not the only significant addition in Cubase 13: Steinberg also added the new VocalChain plug-in. While this essentially combines the facilities offered by a number of Cubase's existing plug-ins, it's impressive just how quickly it lets you go from raw vocal to a polished mix-ready sound. So, with a few vocal examples at hand (you can listen on the *SOS* website: https://sosm.ag/cubase-0524), let's explore the possibilities.

VocalChain: all you need to add polish to your vocals in a single plug-in.

Go With The Flow

The main screen shows VocalChain in action. Arranged down the left edge is the full set of processing modules offered, and these are arranged as three sections, Clean, Character and Send. Your audio is processed through these in order to apply 'corrective' processing, add character/sonic flavour and then ambience/stereo imaging. Individual modules can be engaged or bypassed as required and, in a section, you can change the order of individual modules (drag a module up/down to reposition it). With a total of 16 modules (you can use them all if you need to), this is quite a toolkit. But because it loads as a single plug-in and everything's available in a single window, it's very easy to navigate.

The GUI provides three different levels of control. In the screenshot, the Overview tab is selected (top-left, highlighted in yellow), and beneath the spectrum display you then get access to the most significant parameter from each module. However, select the Clean, Character or Send tabs (when selected, these are highlighted in blue, cyan and green, respectively), and the choice of controls changes to focus on the modules in this section, with more control over specific modules. Finally, select an individual module, and the display changes again to provide access to the full control set for that module. It's a clever bit of design that means you can quickly switch between different levels of editing.

There's also a set of style/genre-based presets to get you started, and these should not to be underestimated. OK, so there's no AI involved here (VocalChain doesn't listen to your audio and then make some setting suggestions in the way that, say, iZotope's Nectar might) but they're well worth exploring and can get you off to a flying start. You just find a preset that provides a suitable starting point and then tweak to taste, using any of the three control levels described above.

The Pitch module does automatic pitch correction but can also be used to fake a vocal double.

Time To Tweak

In terms of that tweaking, a sensible initial task is to use the input and output metering on the right to set your levels. Setting the input level control to get your signal into the green coloured range of the meter is a good start, as it will most likely ensure your signal hits the first active dynamics stage in the preset's design at an appropriate level. You can then adjust the output level to find the happy place where the vocal sits most comfortably in the mix. It's also interesting to watch the two meters during playback: with two compression stages, two dynamic filters and two de-essers available, there can be a serious amount of dynamics management going on, should you need it.

While tweaking, one further feature makes it much easier to evaluate the impact of the changes you are making: the ability to solo each module. In the list of modules, this solo mode can be activated via the small 's' button located to the left of each module's name. Once activated, all other modules are bypassed (so the overall signal level might also change),

but it allows you to more easily focus on what the current module is doing to your vocal's sound. And, by also using the selected module's bypass button, you can easily assess the impact the module is having on the unprocessed signal. These auditioning options are particularly useful for the various EQ, dynamics, filter and exciter/saturation modules.

Make It Pop

So, what about the processing itself? As I said, there's a lot packed in here, but a few highlights can serve as examples — remember to check out the audio examples on the *SOS* website if you want to hear some of these options in context.

Let's start with a 'pop' vocal example. VocalChain includes a number of suitable presets, such as Perfect Pop Dry Vocal or Shiny Pop Vocal, that can deliver a very crisp and compressed, if (deliberately) not particularly natural starting point. Another common pop production technique is to add some 'weight' to a lead vocal by blending in a vocal double an octave below the main sung line, and the Lead Vocal Reinforce preset does just that. While it also provides dynamics and EQ settings that are suitable for modern pop, the 'weight' is added using the Pitch module. As shown in the screenshot, this can be used to apply some automatic pitch correction (either subtle or not so subtle — try the Trap Icon preset), but that's not being used here. Instead, this preset uses the Detune and Formant controls to pitch-shift the vocal down by an octave, along with a suitable downward shift of the formants that makes this down-pitched voice sound a little more natural. Finally, the Mix control has been used to set the blend of the original voice and the 'octave down' version. So it's the same vocal, but with more 'weight'.

Rock On

Lots of rock or metal singers can achieve aggressive vocal distortion through their singing technique, but this is also something you can enhance or create through processing. Here, the Hot Rock Hot Valve Mic Chain preset does just that. While the Character section's Exciter module contributes, it's the Saturator module that does the heavy lifting.

As shown in the screenshot (and can be heard in the audio examples), using the Distortion mode and the Drive control maxed out, this preset doesn't hold back, but it illustrates what's possible.

The Filter Bank feature in the Saturator module lets you target the specific frequency range for any distortion.

It's also worth noting that the Filter Bank is engaged — this focuses the distortion in the 500Hz-3.5kHz region. It's a very useful option and, in this case, it enhances the gritty, lo-fi nature of the sound. If you want to dial it back a bit, then Tape and Tube modes, and different Drive and Mix settings, make that easy. And, of course, all these controls can be automated in Cubase if you want to add that saturated edge just to specific words or phrases in the performance.

Duck Duck Go

The benefit of the Send section is that it avoids the temptation of sending your lead vocal to reverb or delay effects used for more general duties in your project and, instead, you can configure settings specifically for the vocal part. In busy mixes (for example, an uptempo EDM project), too much delay or reverb can easily muddy a mix. However, as the Platinum Female Vocal Chain preset illustrates, VocalChain's toolset allows you to manage this while still getting epic with your vocal ambience.

As shown for the Delay module in the final screenshot, two particular features are useful. First, as with the Saturator module, both the delay and reverb modules offer a Filter Bank, allowing you to trim out frequencies in the delay repeats (or reverb) so you don't get excessive low mids (to clog up the mix) or (at the top end) repeats fighting with your hi-hats. However, it's the ducker's Amount and Release controls that are the stars of the show. They allowing you to suppress the level of the delay (or reverb) while the source vocal is present, and then control how quickly that ducking is released (so you hear the delay in all its glory) between the vocal phrases. It's a classic trick, and VocalChain makes it very easy to pull off.

Join The Chain Gang

There are plenty of very capable third-party 'vocal signal chain' plug-ins designed to tackle the same task, including some powerful ones that feature AI assistance. But until AI can read our minds, it can't know exactly what kind of sound we're trying to create, so there is always going to be project-specific tweaking to be done. Arguably, VocalChain's presets can provide just as valuable a starting point as many AI plug-ins, and because the GUI makes it really easy to adjust every component in a single window, it's super easy to tweak your vocal sound to suit the mix. Of course, the potential of getting quick results is only one aspect of using VocalChain. There's a lot more to explore in the plug-in, so it's a topic I'll probably return to in a future column.

The Delay module also features a Filter Bank, but this feature really shines in the Ducker, where it can help prevent your ambience effects from adding clutter to a busy mix.

Spotlight

Workstation Synthesizers

Looking for an all-in-one playing and sequencing solution? Look no further...

Korg Nautilus

LUKE WOOD

With the power of modern laptops and abundance of virtual instruments and sample libraries, it's never been easier to carry around an entire studio setup. However, there are still plenty of situations where a standalone hardware workstation can be an attractive option. For live performances, having a single instrument loaded up with all of the sounds you need throughout your setlist can be a game-changer, and they can offer a convenient way to get ideas down quickly when inspiration strikes. In this month's Spotlight, we take a look at a selection of instruments that pack in all of the sounds and sequencing capabilities you need to create a song without firing up your DAW.

Akai MPC Key 61 / 37

With the MPC Key 61 and 37, Akai have paired the sampling and performance capabilities of their legendary MPC units with a powerful synthesizer, as well as throwing in a healthy selection of connectivity and interfacing for good measure. Onboard plug-ins offer a wide variety of acoustic and electronic instrument sounds, with rhythmic programming taken care of by 16 velocity-sensitive drum pads with aftertouch, while a pair of mic preamps make it possible to record external sources directly to the unit's internal storage. The unit boasts 128 MIDI tracks that can be used to sequence internal or external instruments, and eight audio tracks that can be loaded up with a wide variety of built-in processing plug-ins. A large multi-touch display is paired with a set of four assignable Q-Link encoders, providing users with precise hands-on control over everything from plug-in parameters to audio and MIDI editing. There's no shortage of I/O, with four line-level outputs joined by a pair of XLR/TRS combo sockets that will accept mic or line-level signals, as well as MIDI in, out and thru connections, eight CV/gate outputs for integrating modular rigs, and built-in USB audio and MIDI interface capabilities. If that's not enough, the MPC Key 61 also boasts USB ports that will accept USB MIDI devices and class-compliant audio interfaces (with support for up to 32 inputs and outputs). The more recently released MPC Key 37 offers the same processing power, and still features the full MPC sampling experience, but in a more compact footprint with slightly reduced connectivity.

Akai MPC Key 61

$ MPC Key 61: $1499. MPC Key 37: $899.
W www.soundonsound.com/reviews/akai-mpc-key-61
W www.akaipro.com/mpc-key-61.html
W www.akaipro.com/mpc-key-37.html

Casio WK Series

There are two workstations in Casio's WK series: the WK-7600 and WK-6600. Both are equipped with a 76-note keybed, but differ slightly in the amount of additional features they provide. A 16-track sequencer with built-in editing capabilities is present on both units, and the WK-7600 is also capable of recording audio from its mic/instrument input; the WK-6600 is still equipped with an external input that can be routed to the output or built-in speakers, but doesn't provide any recording facilities. The WK-7600 comes loaded with 820 onboard sounds and offers 64-voice polyphony, while the WK-6600 provides 700 sounds and 48 voices, and each model is equipped with 260 and 210 built-in rhythms and patterns, respectively. They both offer the same collection of effects including reverbs, choruses, EQs and more, along with an auto-harmonise feature and an arpeggiator, although the WK-7600's top

Casio WK-6600

panel sports an expanded set of hands-on parameter controls.

$ WK-6600 $299, WK-7600 $449.
W www.casio.com/intl/electronic-musical-instruments/product.WK-7600
W www.casio.com/intl/electronic-musical-instruments/product.WK-6600

Korg Kross 2

Korg's Kross 2 retains the compact design of its predecessor, but extends the polyphony to 120 voices and comes packed with over 1000 onboard sounds that range from acoustic and electric pianos to strings, drum kits and contemporary sounds aimed at EDM production. Additional stereo samples up to 14 seconds long can be captured via the unit's line input, before being trimmed, normalised or resampled and assigned to one of 16 playable pads. Thanks to Korg's EDS-i sound engine, the instrument also offers five insert and two master effects that can be used simultaneously, with a generous selection of processors offering everything from delays and reverbs to amp modelling and vintage effects emulations — there's also a vocoder that can be used to process the Kross 2's mic input. A 16-track MIDI sequencer allows users to record their keyboard and pad performance along with

Korg Kross 2 61-MB

any controller movements, and the pads double up as a 64-step sequencer. Some 772 preset drum patterns provide a wealth of rhythmic backing options, and an arpeggiator makes quick work of generating phrases or emulating strumming patterns. 61- and 88-key models are available, both of which boast a lightweight design and will even run on AA batteries for the ultimate portable writing solution. If you do want to use the Kross 2 alongside a computer, USB MIDI and audio connectivity makes it possible to integrate the instrument with a DAW, or benefit from additional sounds, patch editing and backing track playback using Korg's range of additional software applications.

$ Kross 2-61-MB $929.99, Kross 2-88-MB $1299.99.
W www.soundonsound.com/reviews/korg-kross-2
W www.korg.com/uk/products/synthesizers/kross2

Korg Nautilus/Nautilus AT

Korg's flagship workstation boasts a huge selection of sounds, with no fewer than nine dedicated sound engines offering acoustic and electric pianos, tonewheel organs, guitars, basses, drums, percussion and more, along with an array of synth sounds provided by the company's MOD-7, PolysixEX, MS-20EX and STR-1 engines. The instrument's recording section features 16 MIDI tracks and benefits from an RPPR (Realtime Pattern Play/Recording) mode, along with 16 audio tracks capable of simultaneously capturing up to four 16- or 24-bit audio tracks at 48kHz. Basic onboard editing functions are provided and it's also possible to automate the internal mixer. There's plenty of built-in processing, too: a separate three-band EQ is available for every timbre, sequencer track and audio track, and there are 12 insert effects that can be assigned to individual or multiple sources, along with four additional effects slots (two send-based, and two for the final output). A set of encoders and buttons provide hands-on control over key parameters, and are joined by an eight-inch TouchView display that offers in-depth control over all of the instrument's functionality as well as providing access to menu systems and displaying detailed visual feedback. There are 61-, 73- and 88-key models available, with the first two equipped with synth-style keys and the third fitted with a weighted hammer-action keybed. The more recent AT variants are equipped with 61- and 88-key aftertouch-capable keybeds (there is no 73-key AT version) and a modified sound library, greatly extending the instruments' expressive capabilities. Korg also offer an official upgrade service for those wishing to add the functionality to their existing Nautilus.

$ $1699.99 – $2899.99.
W www.soundonsound.com/reviews/korg-nautilus
W www.korg.com/uk/products/synthesizers/nautilus

Kurzweil K2700

The latest iteration of Kurzweil's K2 series workstation offers over five times the polyphony of its predecessor, with the company's VAST (Variable Architecture Synthesis Technology) engine boasting a huge 256 voices. There's no shortage of onboard sounds either: a 4.5GB factory library packs in everything from pianos to orchestral instruments, and there's an additional 3.5GB of space for users to populate with their own custom samples, as well as a pair of mic/line inputs for integrating external sound sources. As for synth sounds, a built-in six-operator FM engine is joined by a virtual analogue engine sourced from Kurzweil's VA1 instrument, and realistic tonewheel organ sounds are on offer courtesy of the KB3 ToneReal engine. Thirty-two effects units offer everything from reverbs and delays to modulation and rotary cabinet simulations, and there's a global master effects section

Kurzweil K2700

kitted out with three-band EQ and compression. The K2700 features an 88-note hammer-action keybed that offers up to 16 independent zones, with faders, encoders, buttons, pad triggers and a ribbon controller providing a wealth of hands-on parameter control, and a set of four pedal inputs make it possible to add up to four footswitches and a pair of assignable CC pedals. A built-in 16-track sequencer offers event- and track-based editing tools such as quantise, swing, controller scaling and more, and there's also an onboard arpeggiator and riff generator along with a MIDI CC step sequencer for creating complex modulations. DAW integration is provided by a USB audio/MIDI interface, and USB host functionality makes it possible to expand the onboard control facilities with additional keyboard or fader/encoder units.

$ $2999.
W www.soundonsound.com/reviews/kurzweil-k2700
W www.kurzweil.com/workstation_synthesizers

Kurzweil PC4 Series

Kurzweil's PC4 and PC4-7 are 88- and 76-key workstations that offer many of the features of the flagship K2700 at a lower price. Polyphony still stands at 256 voices, but with the factory sample library (and user sample space) reduced to 2GB, although you still get the full complement of sound engines, onboard effects and 16-track sequencing capabilities. There's a reduction in audio I/O and external pedal connectivity, and although the audio interface capabilities »

» of the K2700 are omitted, the PC4 is still equipped with USB MIDI. The latest addition to the range, the PC4 SE, lowers the price even further while still packing in plenty of features, albeit with fewer top-panel controls. You still get an 88-key hammer-action keybed, 256-voice polyphony and a 2GB factory library, but with five split zones and no additional sample support. The FM, VA1 and KB3 engines remain, as does the 16-track sequencer, and there are still the same amount of onboard effects — although with more limited editing options.

Kurzweil PC4

$ $1699 – $2499
W www.soundonsound.com/reviews/kurzweil-pc4
W www.kurzweil.com/workstation_synthesizers

Roland Fantom

The Roland Fantom series comes loaded with a vast array of onboard sounds, with their V-Piano and SuperNATURAL technologies promising to deliver the most realistic piano playing experience possible. Synth sounds are taken care of by the company's ZEN-Core engine, and support for their ACB (Analog Circuit Behaviour) technology — and the recreations of iconic instruments it brings with it — can be added with an optional Fantom EX upgrade. Internal and external sound sources can be sampled directly to built-in pads for triggering, or assigned to the keyboard to create custom pitched instruments. As for sequencing, there are 16 MIDI tracks that can each house up to eight patterns, and there's a whole host of processing and effects modules that include EQ, compression, delays, reverbs, chorus and more. Top-panel faders and encoders provide control over key parameters, and more detailed editing and navigation can be carried out on a large central touchscreen. Along with a healthy selection of audio, MIDI and CV I/O, the Fantom features a built-in USB audio/MIDI interface, making it possible to integrate the instrument with a DAW setup, and it's even possible to layer soft synths with the internal sounds and route them through the onboard effects and filters. Roland also offer the more compact and lightweight Fantom-0 series, which deliver much of the same functionality, but with a reduced amount of processing power and fewer onboard sounds.

Roland Fantom 7

$ Fantom $2999.99 – $3999.99, Fantom-0 $1899.99 – $2149.99.
W www.soundonsound.com/reviews/roland-fantom
W www.soundonsound.com/reviews/roland-fantom-0
W www.roland.com/global/products/fantom_series/
W www.roland.com/global/promos/fantom-0_series

Yamaha MODX+

The three models in Yamaha's MODX+ range all offer the same set of features, and differ only in their keybeds: the MODX6+ and MODX7+ are equipped with 61- and 76-note semi-weighted options respectively, while the MODX8+ sports an 88-key GHS (Graded Hammer Standard) action. A 128-voice AWM2 (Advanced Wave Memory 2) engine derived from the flagship Montage M series offers everything from acoustic instrument and drum sounds to a wealth of synths. Each of the instrument's 16 AWM2 parts boasts 18 filter types, and these are joined by a collection of envelope generators, nine LFOs, a three-band EQ and dual insert effects. Even more synth options are provided thanks to a 128-voice FM-X engine that builds on the capabilities and sounds of the iconic DX7, and there's a generous supply of onboard insert and master effects that can be applied not only to the built-in sounds, but also to external sources thanks to a stereo analogue input. A 16-track sequencer capable of storing up to 128 songs is present, and offers real-time replace, overdub and punch-in/out recording modes. Hands-on control is provided by a range of faders, encoders and buttons, along with a Super Knob that makes it possible to simultaneously manipulate up to 128 parameters. There's also a built-in 4-in/10-out USB audio interface in case you do want to the use the MODX+ alongside a DAW, and this supports iOS devices as well as the usual desktop platforms.

Yamaha MODX6+

$ $1349.99 – $1999.99.
W usa.yamaha.com/products/music_production/synthesizers/modxplus

Yamaha Montage M Series

The latest generation of Yamaha's flagship workstation range offers three models: the Montage M6, M7 and M8x. The first two feature 61- and 76-note FSX keybeds with channel aftertouch, and the third is kitted out with an 88-note GEX version with polyphonic aftertouch. Polyphony sits at a staggering 400 voices, with 256- and 128-voice AWM2 and FM-X engines joined by a 16-voice AN-X engine designed to accurately recreate a range of classic analogue synth sounds. The company have overhauled the user interface and implemented a category system that makes searching for and loading sounds quicker than ever, and the collection of hardware faders, encoders and buttons are complemented by a large touchscreen display that offers more in-depth editing capabilities. Sixteen-track sequencing with real-time replace, overdub and punch-in/out recording modes is present once again, and DAW connectivity is provided via a built-in USB MIDI and 6-in/32-out audio interface. The company have recently released ESP (Expanded Softsynth Plugin), a plug-in which replicates the Montage's features inside of a DAW, allowing registered users to work on Montage M Performances without their hardware. It currently offers the instrument's sounds with limited editing capabilities, and Yamaha say that the full version will be available by summer 2024.

Yamaha Montage M8x

$ $3999 – $4999
W www.yamaha.com

SUBSCRIBE & SAVE

TABLET | WEB | REPLICA | PRINT

PLUS BONUS PDF

1 ICONIC MAGAZINE

6 GREAT WAYS TO READ IT

SCAN ME TO SUBSCRIBE

TABLET - PHONE - WEB - PDF - REPLICA - PRINT

All this for less than the price of a Coffee+Muffin each month!

www.soundonsound.com/subscribe

- Go **DIGITAL** - get 12 monthly Tablet issues + all locked Web articles + monthly Full Issue PDF download the day the issue goes live.
- Go **PRINT+DIGITAL** - get the best of both worlds and save over 50%.*

iOS / ANDROID

Every Digital sub includes a 5-device licence to our App: mix and match across iOS/Android Tablet and Smartphone.

ONLY READING SOS IN PRINT?

Choose our full Print+Digital bundle for unlimited 24/7 access.

** The 50% discount offer is based on the US retail price of the Print magazine, Tablet and Web subscription when sold separately.*
Subscriptions can be cancelled at any time and you will receive a pro-rata refund for all outstanding issues.

Guide Parts

Alan Moulder: Why Mentors Matter

If there's one thing that engineers and producers need above all, it's a good mentor and role model — and they don't come much better than MPG Icon Alan Moulder.

SAM INGLIS

Alan Moulder has enjoyed a remarkable career as engineer, producer and studio owner, and a more fitting recipient of the 2024 Icon Award from the Music Producers Guild is hard to imagine. He and producer Flood recently stepped away from running Battery Studios in Willesden after more than 20 years. Now operating from a smaller space in North-West London, Alan is still as busy as ever, these days concentrating mainly on mixing.

Alan's industry recognition is, of course, largely down to his body of work, with an incredible list of credits that includes My Bloody Valentine, Nine Inch Nails, the Killers, Arctic Monkeys, Led Zeppelin and many, many more. However, the award also acknowledges another facet of his career: his role in mentoring newer stars such as Catherine Marks, Adam 'Cecil' Bartlett, Caesar Edmunds and Andy Savours.

The Golden Generation

Like most engineers and producers of his generation, Alan himself learned at

Alan Moulder's current mix room is designed around his distinctive hybrid approach: he still uses a lot of analogue outboard, which is printed back into Pro Tools.

the feet of established figures. "I started at Trident Studios in 1983," he recalls. "The original Trident was owned by the Sheffield brothers and it was very posh. In the '70s it did a lot of Bowie, Elton John and other big records. Then It was sold to one of the engineers, Stephen Stewart Short, and a guy called JP Iliesco, who was a publisher, and Rusty Egan, who was the drummer in Visage. Stephen was the engineer and he trained us, and he was probably the most naturally gifted engineer I've ever come across. But he was a taskmaster.

"I was there for four years, and at the time I was there, Flood was the head engineer, Spike Stent was there, Cenzo Townshend was there, Steve Osborne was there, a programmer called Andy Wright was there, Adrian Bushby was there. We'd all work together, but Stephen was the main guy who trained you. And it was a baptism of fire! If he saw any potential, he'd take you on to be his assistant, and then, a little later, his engineer. We used to call it tour of duty, because some didn't come back!

"And it was long hours in those days. You'd work lots of 24, 48-hour sessions. My first day, I turned up at nine o'clock in the morning and left nine o'clock in the morning. And it was tough, but it was really good fun. You could treat people a lot differently in those days, but I don't regret any of it. It toughened me up."

In At The Deep End

The training that Alan received at Trident wasn't for the faint-hearted, but it enabled him to move into the hot seat surprisingly quickly. "I engineered my first session after, I think, seven weeks, just because somebody turned up and didn't have an engineer. I was then doing a freelance album after seven months. It was a jazz album, produced by Martin Hales. And the great thing was, Martin was a good engineer, so he really wanted a glorified assistant so he could listen. If I got into anything that was over my head he could help me out.

"Although it was very competitive, people were always very willing to help you out. Nobody would ever want to see you sink. If you made a mistake, it would travel around the studio like wildfire and everybody would be laughing at you — but they'd do anything they could to save it. And of course there was no Pro Tools, so when you were dropping in to record, if you didn't get it right, there was no Apple+Z. It was gone. So, it was more nerve-racking in that way, but also a lot simpler in that you didn't have to do all the file backups and things like that.

"If you were working with one of the studio engineers then it was their job to train you. But if you were working with an outside engineer, a freelancer, it wasn't their job to train you, so you'd have a different dynamic with them, and because they didn't know the studio you would help them navigate their way around it. And some were helpful, some were brilliant, some were terrible. You could learn a lot from bad engineers how not to do things. You'd make mental notes: I'm never doing that! I can't remember the names of any of the bad ones, but I remember I worked with an American mixer called John Potoker who was very inspiring to me."

In The Wild

After four years engineering at Trident, Moulder found himself moving into writing and production. Here, role »

Although most of Alan's work these days is mixing, he keeps plenty of guitar effects and amps at hand.

» models were less easy to find. "Martin Hales was probably my first mentor in terms of production, but I ended up then just jumping in. One of the first things I produced was my wife Toni Halliday's solo record, which I produced with her. So that was a foray into kind of writing and producing. I was doing a lot of dance music, but then I started doing more indie, or alternative as it is now. And that's when I started getting more and more work."

Already something of a mix specialist even early in his career, Alan Moulder received in-house training on Trident's SSL consoles. But the '80s were also a time of rapid change, and when it came to MIDI and other newer technologies, there was no option but to dive in head first. "You'd have to learn yourself. When the Atari ST came out, programming was obviously the way things were going. As soon as my wife got her deal and was doing her solo record, I had to learn it. It was a great opportunity to take time off and learn it well, writing, rather than under the pressure of being on the session. Because you didn't have a lot of time. You were working all the time. That's how I got into using samplers.

"My first time with Pro Tools, I think, was on *The Downward Spiral* with Trent Reznor. And that guy knows how to use a computer. Working with them, suddenly it was like: wow, the bar's gone massively up. And I came back and straight away bought a Mac, and it was the four-channel Pro Tools of the time."

One of the consequences of going freelance was that Moulder no longer felt part of an institutional structure where knowledge was shared. "At Trident, your job was to train the assistants there, but then when I went freelance, you'd just be a jobbing engineer going around studios. And it wasn't your job then to train them. You didn't want to overstep the mark in terms of what was appropriate or what wasn't. But then Flood and I set up Assault & Battery Studios. And then suddenly you got your assistant..."

Battery Power

Recruiting assistants at Assault & Battery — now known simply as Battery — wasn't an altruistic endeavour. They were needed to make the studio function, and initially, Alan says, little thought was put into how best to train them. As it turned out, though, the contrasting styles of Moulder and Flood created an excellent

Assisting At The Mix

Mention of mentoring in a studio context calls to mind tracking sessions, with the put-upon runner or assistant rigging mics, coiling cables and catering to the whims of the musicians. But Alan Moulder finds it equally valuable to have an assistant during mixing. "I was mainly the mix assistant at Trident, I did much more than I did tracking. And mixing, in those days, was more boring, in a way. It was all work at the beginning setting it up, then it was making tea and coffee and fetching sandwiches until four o'clock in the morning, when it was time to print and do the recall. Whereas now, you are involved more. I can be working on something and give Finn [*Howells, Alan's current engineer*] tasks to do for me on my B rig. Maybe MIDI mapping, editing or timing; even mix tweaks. So I think they actually get more involved now than they used to, but it is different. You don't get the highs that you get tracking, where you're setting things up and things are being created, and you don't get some of the lows where it's just not happening and you're banging your head against the wall thinking what to do. It's a lot more level.

"There's a point during mixing where I just like to sit on the settee and have them drive, so I can hear. I get it to a point where it's probably 80 or 90 percent there, and then it's much better for me to sit away and either dictate what's done, or get their input too. The thing I did learn from Trident was that they gave you a lot of responsibility early on. So, getting assistants to flex their muscles in terms of their ideas and how they hear things is great. It becomes more collaborative. They get more control and more input. It's better for them, and I can learn things as well.

"If there's something I do that I don't normally do, I will draw attention to it and show why I've done it. Or I'll say, listen to this. I'm a great one for A/B, with, without kind of thing. Or like I'll get Finn or whoever I'm working with to sit here and A/B and see if they can hear what it's doing and whether they think it's better or not."

learning environment. “Flood and I are quite different. By then, I was mainly mixing, and Flood was still doing many productions. I have a very focused way of working and Flood’s anything goes — kind of the opposite. So it was great for the assistants to work with both of us and get different ways of doing things.”

Technology has changed to such an extent that, perhaps ironically, the hardest thing for new assistants to get to grips with was the SSL console, with its text-based green-on-black computer display. “It’s really got its own little way. It was quite difficult for people to get their head around the thing, because it wasn’t a computer in the sense that they knew it — although it’s just signal flow, because people don’t really use the automation now on those desks.

“But when it comes to it, at the end of the day, it’s all about your ears and tuning how you hear things. So that hasn’t changed.”

Two generations at Battery Studios: from left, Catherine Marks, Alan Moulder, Flood and Caesar Edmunds.

Getting An Education

In the ’80s, there was almost no formal education available in recording; most studios hired school-leavers and trained them on the job. Today, although there are far fewer studios and far fewer jobs, it seems as though every university in the land offers a degree in Music Technology. On balance, Alan thinks this is a good thing. “When I started, there was only the Tonmeister course at Surrey University. You had to have physics, maths and music A Levels, so there was no chance for me to get in there. So it’s open to many people now, which is good.”

However, he also warns that a degree course alone isn’t enough to equip young engineers for a career in the studio. “If they come to you after they’ve done a course, that shows to me a good attitude, because they’re not coming out of college and thinking, ‘OK, I know it all now.’ They want to take it a bit further and learn, actually in a proper recording studio. Learning in studios, even with bad engineers, you learn different things. It’s teaching you how to behave in a studio, which is very important. You don’t learn that at college: when to speak, when not to speak, the kind of bedside manner that’s best to make a creative environment. Flood said to me: as an engineer, you’re an invited guest, to kind of make the travel go easily without overstepping your mark.

“When you go to a college you will learn from one or two teachers and their ways of doing things. Whereas when you work with lots of different engineers, you learn lots of different techniques, and you pick and choose what you want, and that becomes your sound.”

> Alan Moulder: “All the assistants we had have totally different personalities, but there was a similar thread of attitude.“

Carrying The Torch

The engineers who came through Battery in the 20-plus years in which Flood and Moulder ran it have been a diverse bunch. “Some came straight out of school, some came from colleges; I’ve had a few from LIPA, they always seem really good. Catherine Marks was a qualified architect and she just decided she wanted to work with Flood, and then I took her on after that. John Catlin, he was going to go to LPIA but decided what’s the point if I can bypass that and save myself the debt. So it’s a mix of both.

“All the assistants we had have totally different personalities, but there was a similar thread of attitude. That’s the main thing: their attitude, their desire to learn, their desire to work. You give them a task, and rather than just do it to the minimum, they did more. Just willingness and good attitude really. You can learn all the other stuff.

“I think we did pretty well at Battery,” concludes Alan Moulder, and the track record of its alumni backs him up. “Seeing Catherine now, she’s up for a fistful of Grammys [boygenius, *whose album* the record *she produced, was nominated for seven and won three*]. She’s got a room around the corner here and we still work together on things. I’ve just mixed a Picture Parlour track for her. And Caesar’s on speed dial for any trouble that I have. And he’s very patient! What I’ve given to him, he’s giving me back. We’re all still in touch and there’s still a good camaraderie between us. And it does give me a lot of pride.

“What was great about Battery was the camaraderie and the sharing of ideas. As well as the tracking room and the mix room, there were programming suites, and so everybody would be milling around. You’d bump into people and you’d say, ‘Oh, God, I’m really struggling with this.’ Somebody would say, ‘Have you tried this?’ If you had a computer problem, there was a whole army of people around to get ideas from as to what to do. And the same thing with sonic problems, or somebody would say ‘Try this plug-in,’ or ‘Try this piece of gear.’ It was great for collaborative work. So, as long as you can have a compound where there is a heart, then I think you will get that team of people come out.”

Korg MS Series REVISITED

Korg's MS range contains some *bona fide* classics — and is much more extensive than you might imagine...

ALEX BALL

In late 1977 Korg's senior engineer Fumio Mieda led a plan to create a series of affordable synthesizers that would help make or break the company. Working late nights and even sleeping in the office to keep the wheels turning, the project was completed by the end of the Spring of 1978. In the 46 years since their release, the products in this series have had a significant impact via a broad and surprising array of styles, and one particular family member has become so desirable that Korg have been producing it again since 2014.

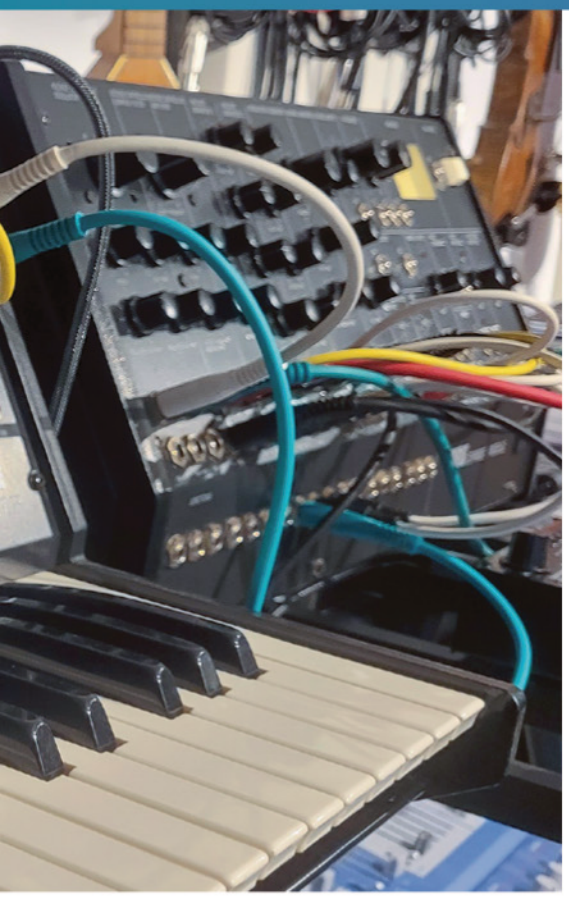
Before The MS

As with most things Korg, the company's inception is unusual. Founder Tsutomu Katoh was not musical and didn't have particular plans to start a music company, but his keen eye for business and willingness to take risks meant that after a fateful encounter, the company was born.

Having served on a submarine during World War II, Katoh subsequently scraped a living with whatever he could get his hands on: car parts, electrical wiring, selling newspapers and in construction. It was whilst he was doing the latter in the 1950s that he was offered a role managing a club, the Minx, in the Kabukicho area of Shinjuku in Tokyo. This role resulted in Katoh being directly in contact with scores of musicians who passed through his club and one particular musician, Tadashi Osanai, approached him with a proposal. Osanai was an accordionist and had discovered the Wurtlizer Sideman rhythm machine, which allowed musicians like himself to carry around a box that provided drums and percussion without needing to hire another player. Osanai believed he could make a better version of the Sideman and so asked Katoh to finance the project. Katoh agreed and in 1963 they set up shop next to the Keio railway line. As their initials were also 'K' and 'O', the name was begging to be used, which is why the company were originally called 'Keio Gijutsu Kenkyujo' ('Keio Research Institute' in English).

After some development, Osanai's design was ready and it was dubbed the 'DA-20 Doncamatic Auto Rhythm Machine', with the name being partly an onomatopoeic reference to the sound of the product; don-ca, don-ca.

In the late '60s, after a series of these rhythm machines, Katoh was approached by another individual with an idea that needed funding. Fumio Mieda already had a track record having invented the legendary Uni-Vibe pedal and also having worked for Teisco. This, perhaps, made the decision easier to



make when Mieda introduced Katoh to his idea to create a new kind of electronic organ. The resulting product was called 'Prototype 1' or 'First Prototype' and, whilst it wasn't called a synthesizer at the time, it contained a monophonic section that was exactly that.

A second keyboard instrument was then developed and it was called the Keio Organ. For reasons that aren't quite clear, a portmanteau of Keio and L'Orgue (French for 'the organ') was used and the instrument was dubbed 'Korgue'. Apparently, due to a typo on some printed circuit boards, this was changed to 'Korg' to match the mistake, rather than get them reprinted! They then used this name for their products before eventually changing the company name itself from Keio to Korg in the 1980s. For simplicity, I'll refer to them as Korg from here on in.

Korg's first production synthesizer was the miniKORG 700 in 1973, and there then followed a prolific five years where a flurry of synths were fired out of their doors; the miniKORG 700S, maxiKORG 800-DV, 900-PS, SB-100, PE-1000, PE-2000, 770, M500, PS-3100, PS-3200 and PS-3300.

By late 1977, Korg were weighing up where to go next and the decision was made to produce an affordable series of compact instruments, with the hope being to tap into the market of potential new synthesizer users. This was, of course, the MS series.

The Arrival

Announced with the strapline 'The Second Generation of Korg Synthesizers', the range contained precisely three of these. The most affordable member of the trio was the monophonic synthesizer (or MS)-10. It basically has one of everything: one oscillator, one filter, one amp, one envelope and one LFO. What makes it more interesting than the simple synthesizer it initially appears to be is that there's a patch panel where the signal path can be reconfigured or interrupted, or where external synthesizers or equipment can be interfaced with the MS-10. Like Korg's earlier PS range, the panel was cleverly placed on the right so that the patch cables wouldn't be in the way of the associated knobs that were on the left.

This instrument was an ideal first synthesizer for fledgling musicians and it was the first synth that Detroit pioneer Juan Atkins owned as a teenager after his grandmother bought him one for Christmas. With its hands-on panel, Atkins taught himself to create every drum sound he could think of, as well as all manner of new effects. His early demos with the MS-10 helped him build a reputation and by 1980, he'd joined forces with Rik Davis to form Cybotron, whose influential electro music was fundamental to the evolution of techno.

The second of the three synthesizers in the series proved to be in the Goldilocks zone in terms of price and functionality. The MS-20 has slowly but surely established itself as one of Korg's most-loved synths, which is evidenced by the fact that there have been nearly a dozen official versions of it, countless clones and emulations and, at the time of writing, you can still go out and buy a brand-new MS-20, 46 years after it was first released!

»

spl **Channel** One *Mk3* The perfect front-end for the modern producer.

Track One *Mk3*
Compact and competent

NEW!

The next generation of Channel-Strips – For more information: www.spl.audio

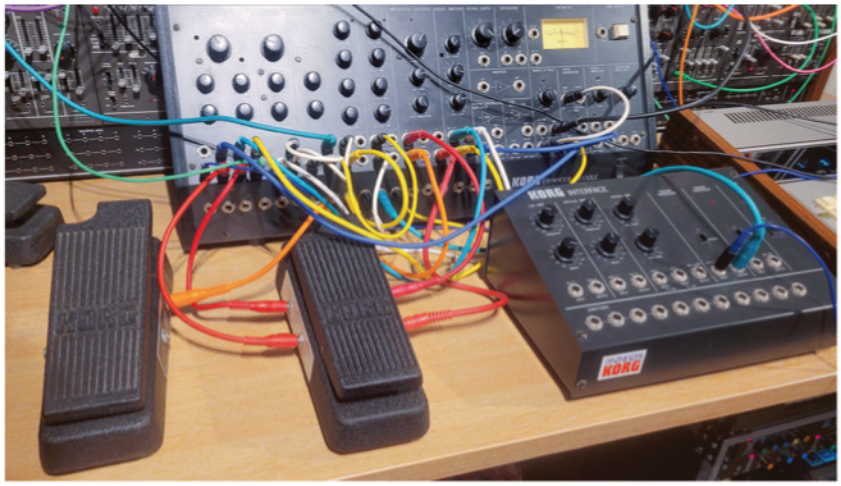
» The MS-20 essentially has twice the functionality of the MS-10: two oscillators, two resonant filters, two amplifiers (the second is in the patch panel), two envelopes, one LFO and a ring modulator. It also sports a more sophisticated patch panel with a dedicated external signal processing section.

A defining part of the MS sound is the filters. On the MS-10 and original MkI MS-20, these were the 'KORG 35' (aka 'Type 35') design that had been introduced with the PS series the year before. Requiring just a handful of transistors and resistors, this was an affordable and compact solution, but it certainly didn't have a cheap sound. The KORG 35's resonance (or 'peak') is very extreme and breaks up and distorts in a fantastic way, resulting in a screaming and growling quality. When two are combined in series and configured as resonant high-pass and resonant low-pass (as they are on the MS-20), all manner of sounds are possible from guttural filth, guitar-like tones, strangely human formants, clangs, bells, womps, belches, squelches and more. This dual HP/LP filter concept was a signature part of the '70s Korg sound, going right back to their earliest prototypes.

The MS-20 also has a clangourous ring modulator tucked into the second oscillator wave selector; the pulse square waves of the two oscillators run into it, regardless of which wave is being used for audio from VCO 1. The patch panel on the right contains some quite sophisticated options, such as a discrete CV input for VCO 2, inverted outputs from the envelope generators, dual wave outputs from the LFO (which could be manually waveshaped), a completely open-ended sample and hold circuit and modulation inputs for the oscillators, filters and amps. The mod wheel and trigger switch on the left of the instrument are not connected to anything by default, which might seem counterintuitive, but the pay off is that their outputs are also found in the patch panel where they can be routed to a variety of destinations.

Finally, the external signal processor was really the icing on the cake. It can be used to overdrive and filter signals, but it can also convert monophonic audio into

Footwork & Black Boxes



The MS range originally contained nine main products, and perhaps the least well-known of the family are the MS-01, MS-02, MS-03 and MS-04. Two of these four units look like Korg branded wah-wah pedals, but are control voltage pedals that interface with their synthy siblings.

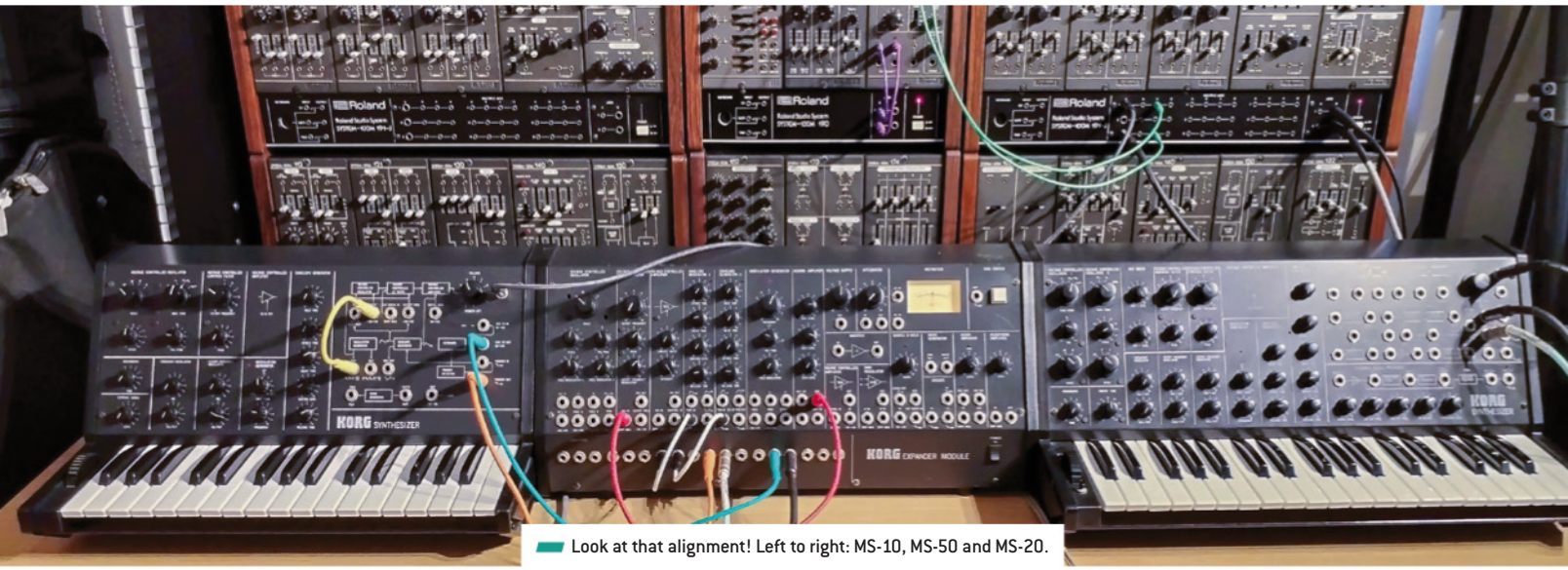
The MS-01 has two functions with associated jack sockets on the respective sides of the pedal; one function is a control voltage, either positive or negative, and the other side utilises the pedal as an attenuator. The MS-04 takes things further with a built-in LFO (with a sample and hold function) and a CV bend, combinations of which are available from two outputs on the opposite side. In fact, these weren't the only pedals that Korg made in these branded chassis; there were also the five 'FK' pedals that included everything from their 'traveler' filters, phaser/wah/double wah, dual volume controls, signal crossfading and more. Quite the offering for those with keen toesies! In fact, Fumio Mieda had cut his teeth building effects pedals (including the legendary Uni-Vibe), so it was probably quite a natural choice to produce these accessories.

With the two MS pedals, all sorts of performance related control could be achieved whilst keeping the players hands free; volume swells, filter sweeps, frequency modulation, glissandi, you name it. So, given the usefulness of these pedals, it's perhaps surprising that relatively few of them seem to have been made and fewer still have survived the decades. Maybe learning synthesis was challenging enough, without needing to involve all four limbs!

By contrast, the MS-02 is a rectangular block that seems perplexing, with terminology like 'log amp', 'antilog amp', 'junctions', 'Vth = 2.5V' and '0V-15V', but it was actually well thought through and very useful. Inspired by the way that an attenuator on an electrical measuring instrument worked, Korg had come up with their own way of controlling the frequency of oscillators across many octaves, which they found made the tuning more accurate and more stable. This standard was dubbed 'Hertz per Volt' and was abbreviated as Hz/V. This was different to the 'Volts-per-octave' (V/oct) standard used by most other manufacturers because that required exponential conversion, whilst this new approach used voltages that corresponded to exponential frequency changes. Put simply, if you plugged a V/oct device into a Hz/V device (or *vice versa*), this would result in a strange temperament and the notes would get further and further out of tune as the player moved up and down the keyboard. Aside from experimental music, this generally wasn't very useful and so the MS-02 converted either to the other so that equipment from different manufacturers could play nicely.

However, this wasn't the only thing required to get Korgs to sing along with Rolands, ARPs or Oberheims; there are two ways to fire off envelope generators on a synthesizer, which are crucial to being able to make any sound (or at least, controllable sound). The first way is a 'voltage trigger' where a positive pulse sets the envelope running, with the key-on time defining the gate. The other is essentially the opposite, dropping from a positive voltage down to zero, at which point the envelope generator starts. If the wrong method is used, pressing a key results in silence, whilst releasing a key results in a constant drone. The solution? The trigger processor section found in the MS-02, which flips one to the other. Prior to MIDI, these were the lengths manufacturers had to go to essentially help integrate their products with those made by rivals, but it shows that they already knew that it was important.

The MS-03 is another metal slab with an esoteric purpose. Like the ESP of the MS-20, it converts audio signals into voltages that can control a synthesizer. In essence, this allowed the user to play a synthesizer from a microphone, guitar, saxophone, kazoo or anything you fancied running into it. Like the MS-02, it had cross-brand-friendly outputs with Hz/V, V/oct, S-trig and V-trig. It also had footswitch inputs so that a player could latch a note and have it hold until it was cancelled, plus an envelope was generated from the incoming signal that could be sent on to the synthesizer to shape the sound. This kind of circuit is called an 'envelope follower' and allows the sonic characteristic of one instrument to be applied to another.



Look at that alignment! Left to right: MS-10, MS-50 and MS-20.

control voltages, envelopes and triggers that can be used to play the MS-20 from an instrument or microphone. The thinking was that it would entice guitarists who could use it as an extended effects box for their guitar, but the reality was that it was quite quirky and behaved strangely. This, of course, made it ideal for experimentation.

All this added up to make the MS-20 quite unlike any synth on the market, certainly at the price. As wonderful as they are, the revered Minimoogs and ARP Odysseys of the day couldn't make anything like the range of sounds possible on the MS-20, despite the Korg being much cheaper and more compact.

The final synthesizer in the MS series, the MS-50 was a keyboardless expander for the MS-10 or MS-20. Unlike its siblings, this synthesizer has no pre-wired signal path. Instead, every single part of the synthesizer has jack sockets for inputs and outputs, making it completely open-ended. Given the more specialist nature of the MS-50, fewer were made and they remain a prized possession for the MS aficionado.

The European Connection

Some 9000km away from the factory in which they were built, the MS-20 became the synthesizer of choice for German new wave musicians, particularly when paired with the MS series analogue step sequencer, the SQ-10.

In 1979, one such musician, Kurt Dahlke took the name 'Pyrolator' and released his first album *Inland*. Featuring MS-20 and SQ-10 with an organ and some Italian synths, Dahlke explored the experimental and dystopian possibilities of these (then) new instruments.

Dahlke had also originally been a member of Deutsch Amerikanische Freunschaft (DAF) but had left the group early on. Guitarist Wolfgang Spelmans, bass guitarist Michael Kemner and Dahlke's replacement, Chrislo Haas, also subsequently left the group and the only »

SOUND ON SOUND
THE WORLD'S BEST RECORDING TECHNOLOGY MAGAZINE

FULL ISSUE PDF

Sound On Sound now offers our Full Issue PDF. This complete digital replica of the Print magazine includes all articles and adverts published in that edition.

Buy and download instantly - no shipping costs!
Only $5.99 each edition
(or FREE with all Digital subscriptions).

Get your FREE PDF today! **https://sosm.ag/FreePDF**

Some sequencers have inspired entire genres of music. The SQ-10 is one of them.

consistent members, Gabi Delgado (vocals) and Robert Görl (drums and electronics) found themselves as a duo by late 1980. Rather than recruit new members, they decided to record their third album (*Alles Ist Gut*) as a pair. Teaming up with producer Conny Plank, the band took to the studio with a minimal approach of synthesizer, sequencer, drums and vocals. Plank processed the sound of the synthesizers by playing them back through amps and speakers and miking them up, adding to the raw energy of the recording and finding a perfect match for Görl's pounding drums and Delgado's provocative vocals. Along with the use of Plank's ARP Odyssey and Sequencer, the duo were ongoing users of their MS-20 and SQ-10 sequencer.

Another former member of DAF also had an enduring relationship with the Korg MS series. Chrislo Haas had teamed up with fellow German Beate Bartel to record as CHBB in 1981, and photos of their studio at the time reveal that the MS series was pivotal to their sound with them, at one time, owning four MS-20s, two SQ-10 sequencers and an MS-50 expander. That same year, the duo met Krishna Goineau, who was only 17 at the time. The three formed the trio Liaisons Dangereuses, with Haas and Bartel's driving, sequenced electronics providing the background to Goineau's chanted vocals. Whilst they only made one album as a trio, they helped pioneer 'Electronic Body Music' and their song 'Los Niños Del Parque' remains a cult classic. The 24 steps of the SQ-10 sequencer give the song its unusual six-beat groove, with the MS synthesizers providing the sounds.

> "As new genres emerged, the old MS synthesizers proved ideal instruments for the job."

Meanwhile, over the border in France, another young duo were making use of MS synthesizers to help define their sound. Daniel Favre (aka 'Spatsz') and Mona Soyoc founded KaS Product in 1980 and were part of the cold wave scene of the late '70s and early '80s. Their best-known song 'Never Come Back' features frantic Korg MS-20 and Moog Prodigy with drum machine and profanity-laced vocals. The video for the song shows close-ups of the unmistakable plastic side panels of the Korg MS series as the duo perform in an abandoned warehouse.

Whilst you could argue that these acts used the MS products because that's what was available to them, the unmistakable sound of the MS instruments, particularly the filters, meant that they informed the sound of the genres as much as the artists. Laden with knobs, wires, blinking red LEDs and phosphorescent voltmeters, the punky but sci-fi-black panels of the Korgs also looked very much the part next to the denim, leather and barnets of the bands that used them.

Talk To Me

Maybe the most distinctive member of the MS series is the VC-10 vocoder. Housed in the same basic chassis as the MS-10, it sports a glowing VU and Korg-branded gooseneck microphone. This vocoder may have also been the first with a built-in keyboard. Combined with internal tone generators and the ready-to-go mic, it is very quick and convenient to use.

At a similar time to recording Yellow Magic Orchestra's debut album, Ryuichi Sakamoto was working on his debut solo album *Thousand Knives* in 1978. Sakamoto personally thanked Korg for their "great

SOUND ON SOUND

PODCASTS

Ryuichi Sakamoto

1952 - 2023 A Tribute

As a tribute to Ryuichi Sakamoto on the first anniversary of his passing, Caro C talks to Richard Barbieri, Natalie Beridze and Carsten Nicolai who share their insights into his mindset and methodologies.

Mic Polar Patterns - Part 2

Cardioid, Supercardioid, Figure of 8 and Omni

A valuable test for recording engineers, David Mellor gives examples of different mic types to emphasise the importance of knowing your mic collection in detail.

Afrodeutsche

A Journey Through Music

Afrodeutsche talks to Caro C about her musical journey, from her introduction to the music industry in Manchester, finding her sound, getting signed to Skam Records and becoming a BBC 6 Music DJ with a prime-time Friday evening slot.

Guy Massey - Engineer Producer

The MixBus Interview

Guy Massey talks about his training at Abbey Road, how this gave him the confidence and experience to become freelance, and how he enjoys blending new technology with traditional recording spaces.

Follow our channels by subscribing to the shows on Apple Podcasts, Google Podcasts, Spotify, Amazon Music or wherever you get your podcasts.

Listen on Apple Podcasts · Listen on Google Podcasts · LISTEN ON Spotify · Listen on amazon music

All shows are mastered to the highest quality the podcast channel will support and are in stereo.

Check out our website page for further details

www.soundonsound.com/podcasts

The VC-10 Vocoder, much beloved of Ryuichi Sakamoto.

» cooperation" in the credits of this album and the recording included their PS-3100, SQ-10 and VC-10. The latter has prominent use in the opening of the title track as Sakamoto's yearning vocoded a capella voice introduces both the song and album.

Fellow countryman Isao Tomita also made use of the VC-10 (credited as "Korg Vocoder") on his "musical fantasy of science fiction" album *Bermuda Triangle* in 1979. In fact, the final sounds on the recording are strange whispers running through the VC-10 as the album's journey through vast, deep sonic textures resolves by diminishing back towards silence.

The End Of The Beginning

By 1983 Korg had moved onto new technology and so the MS series was discontinued, but musicians were certainly not done using them. In fact, if anything, the popularity grew after the series was deleted. As new genres emerged, the old MS synthesizers proved ideal instruments for the job and they began cropping up again.

In 1997 William Orbit was brought in to produce Madonna's *Ray Of Light* album and the MS-20 proved an essential part

A Tale Of Giants

One of Korg's many unique stories is that of their synthesizer studio and school that was established in Tokyo around the time that the MS series was completed. The idea was to offer classes teaching synthesis to beginners, with the ulterior motive probably being to sell some more synths in the process! As there were no interactive whiteboards in the late '70s, Korg took a rather novel approach and made a few giant MS-20s for the teacher to use at the front of the class, hence they are known as the 'Blackboard' versions.

At the time of writing, only one example is known to still exist and it belongs to Don Muro. Don was a part-time clinician and performing artist for Univox (the US distributor for Korg) in the 1980s and early '90s and was gifted the giant synth as a thank you for his work on the M1, for which he produced a factory demo song (it's 'SONG 4 Ms. Muro' if you want to track it down in your M1).

Prior to that, it had lived in a portable room that was set up at trade shows to allow for sound separation on the noisy show floors, although the giant MS-20 never travelled with the room.

Korg also later made 30 slightly smaller (but still giant) 'Export' versions with yellow legending that were sent to local offices and distributors around the world. Like the earlier version, these weren't just props, but fully functioning MS-20s! In fact, the very same circuit boards were inside, just with long cables stretching to fit the giant chassis, and disks were created to artificially increase the size of the knobs.

of the production. As well as providing straight synthesizer sounds, the instrument was perfectly suited for creating effects and textures such as those heard throughout 'The Power Of Goodbye'. Orbit has also stated that numerous sounds that people believe to be guitar on the album are actually the MS-20.

One of the best-known uses of the MS-20 was by Quentin Dupieux (aka Mr Oizo) on his 1999 number one single 'Flat Beat'. Sampling his MS-20 into an Akai S1000, Dupieux then sequenced it into an infectious groove with a demo version of Cakewalk Express! Being the soundtrack of a memorable TV commercial featuring a distinctive yellow puppet (who also appeared in the music video) did the song no harm whatsoever and helped lift it to the top of the charts in several countries.

Another song to be helped by being the soundtrack for an advertising campaign was 'Bohemian Like You' by the Dandy Warhols. Originally released in 2000, it was re-released in 2001 after being used by a telecommunications company, whereupon it flew up the charts across Europe in particular. Without a bass guitarist, the band's keyboard player Zia McCabe provides the bottom end with her Korg MS-20. In the video for the song, McCabe can be seen playing the MS-20 during the karaoke bar scenes.

For those willing to dive into the possibilities of the external signal processor section, the audio you give it can be transformed into something unexpected. This proved the ideal tool for Goldfrapp on their song 'Lovely Head' in 2000. The peculiar, theremin-like sound heard in the song was actually produced by Will Gregory processing Alison Goldfrapp's voice through the ESP of his MS-20.

Another German musician with a connection to the MS-20 is Felix Kubin, who went as far as to compose a work for an orchestra of 20 MS-20s called 'A Choir of Wires'!

Supergroup Atoms For Peace were also fans of the MS-20 in more recent years, with Nigel Godrich using the instrument live and in the studio.

Looking back at the decades we can see that the MS series was relevant at the time of its release, but that it became relevant again and again in different contexts as musical fashions changed. The fact that so many big names and genre-defining artists have turned to its family members tells us that the MS series is unquestionably classic.

Da Funk?

Common knowledge can have its uses, but it can also be wrong. For example, for decades it was 'known' that the bass line to Michael Jackson's 'Thriller' was played on a Minimoog. That was until recently when Anthony Marinelli and Greg Phillinganes revealed that it was actually an ARP 2600. Likewise, it's currently 'known' that the main riff to Daft Punk's 'Da Funk' was performed with a Korg MS-20. Now, to be fair, it sounds absolutely like a Korg MS-20 and the instrument can perfectly recreate it, but the band have never confirmed whether it was and to avoid another, erm, thriller, I've not included it in the main article.

FNZ are Finatik aka Michael Mulé (left) and Zac De Boni.

INSIDE TRACK

FNZ: Finatik & Zac De Boni

Hard work and a love of sampling have made FNZ the hottest production duo around.

PAUL TINGEN

"When we started flipping samples in 2018, very few were doing that. We'd send our packs to people, and they'd say, 'Oh, man, can you send me some non-samples, please?' And we were like, 'Sorry, but no, this is what we're doing. Take it or leave it.'

"Since then sampling has made a full-scale comeback. Look at Jack Harlow's 'Lovin On Me', Drake's 'First-Person Shooter' [*both 2023*], and Drake and Future's 'Way 2 Sexy' [*2021*]. Since then everyone has followed suit and is chopping up samples. We love it, because it's the foundation of what we do."

Speaking is Michael Mulé, aka Finatik, one half of FNZ, the other half being Isaac 'Zac' De Boni. The production duo have been involved in hits by Kanye West, Kid Cudi, 21 Savage, Nicki Minaj, Drake, the Kid Laroi, Jack Harlow, Burna Boy, Offset, Travis Scott, Kendrick Lamar, Future and many more, earning three Grammy Awards in the process.

Slow Burn

Except for Kanye, all FNZ's above-mentioned big-name credits date from the current decade, when major success finally hit. With credits dating back to 2009, it means that the duo spent considerable time getting to where they

are now, working hard to improve their skills, and making the right connections.

"For years the stuff we were making wasn't cutting through at the highest level," comments Mulé, "because it wasn't as unique or polished as it should have been. We didn't have an identity yet. The feedback let us know that we weren't ready yet. We weren't getting the song placements we wanted, and we weren't working with the people we wanted to work with. We spent a long time figuring it out. I think it was when we started working with artists like A$AP Rocky [*2012*] that the music we were involved in making began to represent our true soul in terms of experimenting with different colours, different atmospheric sounds and generally sounding different. After that it wasn't until 2018-2019 that we made stuff that when Kanye heard it and Drake heard it, they were like 'Yeah, let me get on that.'"

Sowing The Seeds

Although they are now based in LA, the duo are originally from from Perth, Australia, where their love of sampling was kindled. "For me it started in 1999," recalls Mulé, "when I saw the Beastie Boys with Mix Master Mike on Australian TV. He was going nuts on a Vestax turntable. I was like, who is this guy? I became obsessed with turntablism. I saved up enough to get some turntables and a mixer, and started researching music and collecting records. I got heavily into the DJ battle circuit, with DMC championships, under the name DJ Finatik. This plateaued when I was 16, after which I started making beats. I again saved up money, to buy an Akai MPC2000 and other studio gear, and I studied what people like DJ Premier, the Alchemist and Pete Rock were doing. For a long time I was really bad at making beats!"

De Boni's starting point was different. He played piano, and then, "at the age of 14 or 15, a friend of my brother introduced me to Fruity Loops. I began using it, and made beats for friends. From there I went to Reason and other software, and just kept hacking away at it. I was terrible, but for some reason kept going. When I got a bit better, a mutual friend introduced Mike and I. We lived maybe 15 minutes from each other."

"Zac started coming over to my mum's house," continues Mulé. "I had an MPC4000 by then, and a Digidesign Digi 002, with the mixer, hooked up to Pro Tools. The 4000 was a huge, clunky piece of equipment, but it was great. We had a Yamaha Motif Rack synth as well. We made beats from scratch, often using samples."

California Dreaming

Finatik 'n Zac, as they were known, gradually built a reputation as the best beatmakers in Perth. At one point De Boni also attended the SAE Institute in Perth to sharpen his studio skills. But they had dreams of moving to the US, where, says Mulé, "the music was created that we were fans of. A friend of ours had a show at a small radio station in Perth called Groove FM, and did interviews with big producers from the US. One of the guys he interviewed was Jim Jonsin, and our friend got him to listen to some of our music. Jim said he wanted to sign us, but as time went on, we heard nothing and lost contact.

> Zac De Boni: "Giving space for other producers to do their thing doubles the chances of it getting placed, because our network combines with someone else's network."

"We were doing regular jobs at this point. I worked at my mum's café and Zac at an Italian restaurant. Eventually we realised that if we wanted to make this happen, we had to fly to Miami, and contact Jim as soon as we got there. So we did, with all our equipment, and he was just gobsmacked and shocked that'd we'd flown across the world. This was in the beginning of 2009. Three months later he signed us. We continued travelling up and down between Perth and Miami, and at the end of 2010 we moved permanently.

"Jim would be in his main studio, with artists like Kelly Rowland, Pitbull, Usher, and so on. We could come in, meet them, and then we went to a back room to build our own clientele and repertoire and confidence and this and that. Two years later, around 2011-12, Jim said, 'OK, you guys are ready now,' and he invited us in his room while he worked with Ludacris and A$AP Rocky and others.

"During that time none of the songs we worked on were big hits, so the royalties were not crazy. It wasn't until we moved to LA, in 2016, that we were getting some financial rewards for all the hard labour. Our time in Miami was about cutting our teeth and learning the ropes. We had sessions with amazing artists and this and that, but were still learning. But we knew that at some point we would have to move out to LA to build our career, as opposed to being in the shadow of a big producer."

"A big shout-out to Jim. He made it all happen for us early on," continues De Boni. "But people were going to Miami to work with him, they weren't flying there to work with us. Moving to LA was the beginning of us starting to forge our own identity, and for that reason we shortened our name to FNZ. We worked a lot with Denzel Curry, and also executive produced his albums. Our name started to build from that."

"Coming to LA was like shedding an old skin for us," recalls Mulé, "it was a new beginning. But to be honest, it was really hard. Many producers had told us, 'When you move to LA, we'll get together.' But after we arrived, cricket silence. It's the name of the game. We simply had to continue to prove ourselves, until people would reach out to us.

"Working with Kanye was another big stepping stone. We worked on his unreleased *Yandhi* album, and then, finally, a track we had done with him, 'Everything We Need', was released on his *Jesus Is King* album [*2019*]. We also were involved in the making of three songs of his *Nebuchadnezzar* opera, including the song 'Wash Us In The Blood' [*2020, with Travis Scott*]. After 10 years cutting our teeth, things really started to happen, and snowballed from there."

Joint Effort

The snowballing culminated in an exceptionally successful 2023, with FNZ credits that include Kodak Black, Young Thug, Trippie Redd, Marshmello, Lil Wayne, Offset, Nicki Minaj and many more. FNZ's most notable credits in 2023 include five songs on the Kid Laroi's debut »

» album *The First Time*, as well as Drake's 'First Person Shooter' (featuring J Cole), and Travis Scott's 'Thank God'. Another major hit single FNZ worked on was Future's 'Wait For U' (2022, featuring Drake and Tems), which won a Grammy for Best Melodic Rap Performance in 2023.

FNZ's credits are almost always as co-writers and co-producers, but these can reflect two very distinct approaches: conventional co-writing and co-producing with an artist in the studio, in which they see the production process through until the end; or supplying starting points for other producers and artists to work with, without any further involvement. But sometimes the two approaches overlap, as the duo explain.

"Between sending out folders with tracks that other producers and the artists use, and working more collaboratively, I'd say our work is half and half," explains Mulé. "An artist like Drake, for example, is hard to reach in terms of being in the room with him. But we have amazing relationships with producers like Vinylz, Oz, Tay Keith and guys like that, who have worked with Drake for a long time. So we'll chop up and flip samples and then pass them along to Vinylz or Tay, or whoever it is, and they'll add the drums to something they like and then play that for Drake.

FNZ with the Kid Laroi (front) and Ty Dolla $ign (right).

But with artists like the Kid Laroi or A$AP Rocky we start discussing ideas with them from the start, and we'll either play them samples or Zac will get on

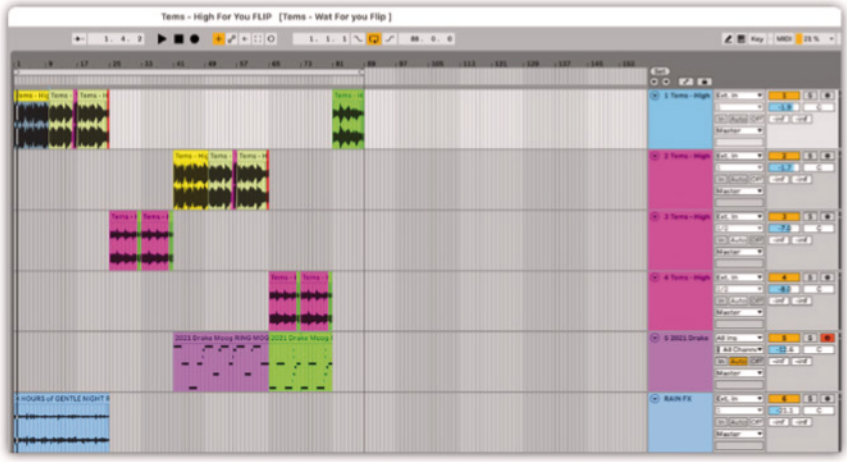
Future 'Wait For U'

'Wait For U', featuring Drake and Tems, was a major, Grammy-winning hit in 2022. It is based on a sample of a track called 'Higher' by singer Tems. Unusually, FNZ sampled a live version performed on Genius, with the singer accompanied by just electric guitar and bass. FNZ's sample session consists of just six tracks: the sample split out over four tracks, a Moog synth, and a rain sample track.

Zac De Boni explains: "The live version had a better vibe, with better sonics, and her performance is better than in the original. Her vocals are a lot more prominent, because it's more stripped-down. I think us sampling a live performance started a trend, because now quite a few people sample live versions!"

Michael Mulé: "We sampled different parts of the song. Because it was live, the first thing we had to do was make sure the sample is in time. You can see the warp marks. Every beat had to land exactly right, so we had the freedom to chop it up freely. We also sped the bpm up quite a bit to 166bpm. Her version is a lot slower."

De Boni: "The top four tracks in the session are all the sample, and the tracks are colour-coded, so we can see what's what. The top track doesn't have any treatments, apart from compression, because it's a live performance. The next section is the post hook or the verse section, and we used Soundtoys MicroShift and EchoBoy, Valhalla Reverb, EQ, and the Ableton multiband and Glue compressors. We have the same plug-ins on the two other sample tracks, with different settings. So it's pretty simple. It's more like we're tidying up the sound, and the MicroShift gives it a nice little phase effect, almost like a flange.



FNZ's Ableton project for 'Wait For U' contains only six elements.

Mulé: "The track called '2021 Drake' contains a Moog sound from Omnisphere that we called 'Drake Moog', hence the track name. This was before we started producing with Drake. The preset we used for the bass sound is 'Moog Modular Big Booty'. We also added some rain to give the track an ambient vibe. You can probably barely hear it in the instrumental, but it adds a nice little touch. The rain sound comes from a sample pack.

"After we had chopped up the sample and added these elements to it, we sent the loop to producer ATL Jacob. We had pitched it up, but he pitched it down again, and added drums. That led to Future getting on it, and obviously Drake later on. We still didn't know what to expect, but when it was released, it shot straight to number one!"

The Kid Laroi ‘What’s The Move’

The core of ‘What’s The Move’ combined two audio samples with soft synth parts.

‘What’s The Move’ (with Future and BabyDrill) was the final single from the Kid Laroi’s debut album *The First Time*, both released at the end of 2023. FNZ’s session for this project consisted of two sample tracks, two keyboard tracks, a Moog bass, three Serato sample tracks and an 808 track.

Michael Mulé: “The sample is not actually a sample in the normal sense. Instead we used an audio clip called ‘In My Car’, which is just a choir and a keyboard, that was sent to us by a producer we work with, Mickey de Grand IV from the band Psychic Mirrors. It’s rather jazzy-sounding, which is the opposite of the final track.”

Zac De Boni: “We treated the clip like a sample. At the top of the session in red are the two tracks with the ‘In My Car’ sample. We sped the clip up from 137 to 138 bpm, pitched it down, chopped it, and then we played some chords and keys over it, to fill it out. The yellow track is the chords in MIDI, and in light pink underneath is the track on which we printed the audio. The sound comes from Output’s Substance. We converted to audio because it gives a clean result, with no overhang in the gaps. We added the Sonic Charge AudioBode plug-in, with a Swedish ’70s TV reverb, and some EQ. Some of our tracks will have millions of plug-ins, but these tracks already sounded great.”

“The Bass Moog is from iZotope’s Iris 2,” continues Mulé. “After that is a sample, spread over three pink tracks, using Serato. After we had added keys and a bass, we thought, ‘Oh, this would be crazy if we could add a phrase or vocal somewhere to make it pop, to take it to the next level.’ So we scrolled through all our dance a cappellas. We ended up using three parts of a ’90s house song by Kariya, ‘Let Me Love You Tonight’. We put the a cappella in Serato, and chopped it up into little pieces.

“The main thing we used are the vocals saying, ‘Don’t you feel it too?’, which is track 7. Track 6 is the vocal sample used as a stutter effect, and track 8 is a vocal we turned into a rim shot. The original is very clean, so we added the Soundtoys Decapitator and MicroShift. We wanted to make it sound a little more distorted. Track 9 is a nicely distorted 808, from Ronny J. We have millions of 808s to choose from, and this one sounded perfect. It has a really low distorted vibe to give it that bottom end that we needed.

“When we got in the studio with Laroi we went through a bunch of ideas, and this is one that we played for him. There’s another co-producer on the song named Dopamine, who added the drums. This was all produced together in the room, also Laroi’s vocals. The only part that was done remotely was by the Parisi brothers, who are in Italy. They did the additional production on the outro and various little things around the song.”

the keys, and that becomes the starting point for new songs.”

Sampling was foundational to the hip-hop genre when it emerged in the early 1980s, but the ways in which samples today are chopped, looped and treated are dramatically different. “The options are crazy now,” elaborates De Boni. “Sometimes we’ll find a sample and chop it up in the traditional way and make it sound amazing. But we can also extract the vocals from a sample and do a whole section where it’s just a cappella vocals, and then use Melodyne on an old ’70s vocal to change the melody, and add vocal harmonies that weren’t in the original. Or we remove the drums from the sample. We add synths, 808s, tons of different effects, change the key, and so on. We can manipulate and bend samples in many different ways. It’s great fun! We really, really love finding samples, and manipulating them and integrating them with cool other things, so they sound like one original thing, and not like a sample to which we have added stuff.”

Doing Flips

FNZ create their sample flips and song ideas in their studio in Los Angeles. “We don’t live that far apart,” says De Boni, “so I’ll go pick up Mike in the morning, and we head to the studio and crank out 15 ideas a day. We work in Ableton, and have a Focusrite Saffire I/O, and JBL LSR6332 and NS10 monitors. We just got the Mackie Big Knob [*monitor controller*]. We also have an upright piano, a Fender Rhodes, a Sequential Circuits Prophet-10, a Mellotron, and some guitars. We have some microphones, but over the past few years we’ve taken to just putting two iPhones on either side of the piano and recording it like that. We use a handclap to synchronise the two phones. The other keyboards are plugged straight into the soundcard.”

“In addition to the JBL and Yamaha NS10s monitors we also have a big KRK 15-inch sub,” adds Mulé. “For the room everything goes nice and loud. The NS10s are crucial because they allow us to fine-tune things. Our ears can get tired and a little burnt on the JBLs at the end of the night, and it’s nice to sometimes work quietly on the NS10s. Some producers love to work loud all the time, but when you turn it down you can focus a little more on detail, and your ears aren’t going to get fried so quickly. When you get it to sound great like this, and then crank it up, it sounds amazing.

“When we were working with Jim in Miami, we were still were using Pro Tools for recording. But for production, we started using other things. We tried Logic, Cubase, Reason, Acid, everything. Then around 2014, DJ Dahi introduced us to Ableton. That made the most sense to us, and we’ve stuck with that ever since. It was a lot more intuitive than Logic for production. The way that you could manipulate audio in Ableton worked far better for our purposes, even at the time.

“It’s the audio warping, stretching, chopping, looping, all of which was and remains better than in other DAWs. Ableton works perfect for us, and it feels like home now. We also have just about every NI Kontakt library, every soft synth VST, every plug-in for effects, anything like that. We’ve collected quite a lot over the years. We’re always finding new VSTs and new plug-ins. It’s an obsessive sick disease at this point! We also have »

The duo's upright piano is normally miked with a pair of iPhones, sync'ed using a handclap!

» an Ableton Push, for drums, chopping samples, and things like that."

Finding Samples

Until not so long ago, the duo's process in their studio was split 50/50 between starting with a sample and starting with a musical idea of their own. But in recent years this has shifted to starting with a sample in more than 70 percent of cases. "The song 'Where Does Your Spirit Go?' from the Laroi album," explains Mulé, "began with Zac playing the piano, and has no samples. There are other bits and pieces that have come out without samples, like the track 'Keep My Spirit Alive' on Kanye's *Donda* album [***2021***], which started with Zac singing and playing keys, which we sampled and flipped. We just go on what we feel in the moment. But recently we're definitely leaning more on the sample side.

"We just love finding really obscure samples and bringing them to the world. An example is this old '70s Douglas Penn song 'Do You Know' that had just 200 listens on YouTube when we found it. We were like, 'This is an incredible song, we need to chop it up and give it to Jack.' We did, and it turned into Jack Harlow's song 'Denver' [***2023***]. That sample is phenomenal.

"For Drake's 'First Person Shooter', Zac and I were digging for samples, looking for the rarest stuff, and we came across 'Look Me In The Eye', by Joe Washington and Wash, from 1975. At the same time Drake was hitting Vinylz up all the time for more beats. Vinylz asked us, and we sent him a pack of maybe 80 or 90 samples. Our Joe Washington sample flip was all the way at the end, with a random name, because we name things whatever. Vinylz added drums, and he sent the beat back to us, saying that Drake had put it on hold.

"A week before the album came out, Tay Keith hit us up, asking for a dark sample. We guessed it was for Drake, and we found this obscure orchestral string sample, 'Redemption' by Snorre Tidemand. We chopped that up, and sent another folder, and the Tidemand sample became the second half of 'First Person Shooter'. So we provided the seeds for both parts of the song, which was pretty cool.

"Another example is 'Die Hard' on Kendrick Lamar's *Mr Morales & The Big Steppers* album [*2022*]. We had been chopping up a million types of samples around the time we were working with

Kanye, and Kadhja Bonet's 'Remember The Rain' sample was one of them. Kanye didn't pick up on it, so we gave it to producer DJ Dahi, with whom we have a great relationship. He worked on it with Baby Keem when they were producing for Kendrick. We heard the finished instrumental the day before Kendrick's album came out, and we were like, 'Wow, this is crazy!'"

Producing Samples

FNZ's aim is to deliver loops that are as finished as possible, needing only drums and vocals, and a final mix, to result in a releasable track. "We're not just looping a sample," says Mulé, "we're flipping it, and finding the best parts, chopping and arranging that, and we drag the sample over different tracks for different treatments. We add music and other things on top and we're doing a lot of processing and a ton of EQ'ing as well, because sometimes when we flip old stuff from the '70s or '80s, when we start pitching and manipulating it, all these different frequencies start popping up.

"Sound selection is the most important thing, especially when we're sampling obscure '70s, '80s or '90s songs. When we use the Prophet or are going through our VST or Kontakt libraries, it's about having the ear to pick the right sound to lay over the top. You want the sound to blend, and not to stick out like a sore thumb. We may add a missing frequency, like a bass or keyboard for low end. You either play along with the bass in the sample or do something different. Sometimes we pick sounds that are from the era of the sample. Arturia Analog Lab is great for retro sounds, for example.

"In fact, it could be any sound, because we usually degrade the sound, using plug-ins, like the Aberrant DSP Digitalis Digital Wasteland plug-in. Sometimes we throw on a plug-in that can take out drums, to get a wishy-washy effect as the transients are smoothed. Sometimes we take out the drums altogether, or we'll extract vocals. Drum removal plug-ins or AI can create annoying artefacts and take away the clarity or purity of the sound, so we tend to edit drums out manually, replacing them with bits from elsewhere in the song or stretching the audio to fill the gap. It can get pretty surgical!"

"The aim often is to make it sound retro and current at the same time," adds De Boni. "Sometimes we'll have a sample that is kind of soft, and we'll put a really hard 808 hitting under it for contrast. We just mess around. We try not to think about it too much from a technical perspective. I'm always making fun of Mike because he'll put five EQs in a row, things that traditionally people wouldn't do. But if it sounds good, why not? Whatever it takes to get the idea to work."

Drop The Drums

The resulting projects are notably minimalist, usually containing fewer than 10 tracks. De Boni: "It's something that we learned a long time ago. When we first started producing and making beats, we had a million tracks, with tons of layers. Over the years we've learned to do a lot less and be more minimal. Every sound and every track that's in our sessions is doing something. We learned not to complicate things. When we find a sample we like, we'll add what we feel it needs, but we don't want to overproduce it."

> Michael Mulé: "We're not just looping a sample. We're flipping it, and finding the best parts, chopping and arranging that, and we drag the sample over different tracks for different treatments."

The final step of FNZ's process is adding drums. However, they then take them off again. De Boni: "Everyone's got their own idea of when something's finished and ready to send out. But for us, we always test with our own drums. If it sounds like the full finished production with the drums, then we know it's ready. We then mute the drums, and bounce the session down. Another advantage of testing it with drums is that it will sometimes highlight timing errors that you might not hear with a metronome.

"Also, just before we bounce something down, we'll throw Waves SoundShifter on at the end on the master, and start pitching the track around, to find the perfect key. We can work on something for two hours in a certain key, and right at the end when we put the drums on to test it, we may change the key at the last minute. We may change the key three semitones down or up, or whatever, and then bang, that's the perfect key. It'd done."

Removing the drums is as much a business decision as a musical one. "A lot of that is to do with networking," notes De Boni. "If we send another producer something that already has drums, they have nothing left to do on it, and they're not going to want to play it for Future or whoever it is. They want to do their part. Giving space for other producers to do their thing doubles the chances of it getting placed, because our network combines with someone else's network. They're moving the beat around. We're moving the beat around. Their publisher is moving it around, and our publisher is moving around. Also, we like to hear other producer's takes on our loops. Drums may be their strong point, whereas our focus is more on the music. So it's a team effort.

"We'll typically send a producer 15 to 20 loops in one pack, mostly sample-based. We sent 80 to Vinylz for Drake because we were trying to put in as much as we could. You never know which one might be the one. The producers are inspired by what we send them and add the drums, either with the artist in the studio or in their own time, and then give the beat to the artist. They don't usually play the beat to the artist until it's completed. There are exceptions, like Kanye, one of the greatest artists of all time who is also a producer — he'll want to hear just the sample loops. Because when there's drums on a sample, it can dictate too much where the song is going to go."

"We're almost OCD when it comes to sending samples out or playing something to someone," concludes Mulé. "We want everything to sound great. We don't want anything to jump out crazy loud or any high end that's going to screech everyone's ears. We always pay attention to detail and make sure something is EQ'ed right and it's got the right compression or processing. We try to get it as good as we possibly can, to where it makes the next person's part easier, and inspires them to take what we have done further."

SAMPLE LIBRARIES

Emergence Audio
Viola Textures
Kontakt Instrument
★★★★★

You could regard Emergence Audio's latest Kontakt Instrument as string section closure, because Viola Textures completes a line-up that already includes Violin, Cello and Double Bass.

If you are a bit of a string newbie, you'd be forgiven for wondering why you might want to plump for this edition, over the other variants in the Textures catalogue. The answer lies with the instrument itself. Despite often being the butt of jokes, the viola's relative size furnishes it with the most amazing sonority. Pitched a fifth lower than a violin, and an octave above a cello, it's lowest notes venture firmly into the bass register. This is traditionally a C, one octave below Middle C, but Emergence have seen fit to extend this to a B, a semitone lower, presumably by detuning the lowest string.

The interface adopts the same design as previous Textures instruments, with 3.6GB of sampled content when installed, all captured at 48kHz/24-bit.

Each patch consists of two sample partials, which are selected from the left- and right-hand side of the instrument, with a drop-down list of articulations providing a selection that is far from the norm. These offer extended phrases of continuous bowing, with interjecting moments that might punctuate, by way of exaggeration in dynamic or playing. The 26 samples are labelled to offer some clues; Ricochet, for example, features the bow dropping on the string before it begins its travel. This is relatively randomised, with no tempo attached. Meanwhile the Normale articulation realises the full travel of the bow, before it reverses direction at each end.

The samples themselves are exceptionally pure and organic, and sound stunning in simple isolation, but it is the additional functionality that brings the movement to the library. Each of the two sample sections have their own set of parameters, such as ADSR for amplitude, high- or low-pass filtering with resonance, panning and expression. Sitting centrally, a large virtual pot allows the blending of the two partials, while an LFO sited below can be deployed in a number of ways.

You can change the LFO's rate and depth, synchronising to your DAW, and route it to the filter, pan and balance controls. This is where you can create real depth and movement, especially with a selection of contrasting samples panned in opposite stereo locations. As the LFO is equipped with five standard waveforms, with control of depth, the range can move from subtle to extreme, very swiftly. You can even create gating effects, using the LFO's square wave.

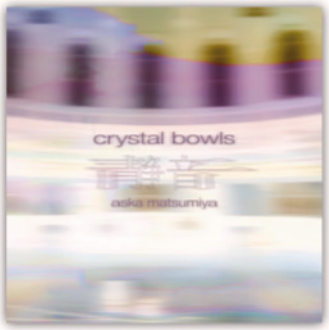
Couple this very musical programmability with an extensive effects section (which includes convolution reverb, delay, and saturation) and you can create everything from scratchy strings to luscious, ethereal pad-like tones.

If you're adding this to other titles in the Emergence Textures line-up, the content is different enough to make it worthwhile. These sounds are very contemporary, making it ideal for media and wider production work. *Dave Gale*

$99

www.emergenceaudio.com

Spitfire Audio
Crystal Bowls By Aska Matsumiya
Kontakt Instrument
★★★★★



Hosted by Kontakt or the free Kontakt Player and weighing in at 2.1GB, Spitfire's Crystal Bowls features seven quartz singing bowls played by composer Aska Matsumiya. The bowls themselves are tuned to the notes of the C Major scale and have been sampled to allow them to be played chromatically. Unusually though, the bowls are tuned to A=432Hz, this purportedly being a 'healing' frequency, though Kontakt's tune control easily brings them back to concert pitch if required. The recording took place in the Hackney Round Chapel, London in order to take advantage of its natural acoustic and to create the instrument's IR-based reverb.

The GUI is straightforward with level controls for each of the six playing types: Brushes, Soft, Sticks, Rubber, Plastic and Hot Rods. These may be mixed, though non-applicable options are greyed out depending on the play mode selected. Above these are controls for attack, release, sample start offset and reverb. Four play modes are available at the bottom of the screen: Shorts, Longs (sustains), (tuning) Forks and Warps. Warps offers sounds processed using guitar pedals and granular treatments, while Forks appears to be the result of touching a vibrating tuning fork against the bowl so all beater options are removed. Depending on which play mode is selected, additional sub-options are shown at the bottom of the screen. For example, select Long and you have further choices of Long, Swells and Rolls. While the sound from a crystal bowl can be close to a sine wave at times, playing with different beaters produces different overtones and nuances that create a unique character.

As you might expect, striking the bowls with various hard and soft beaters produces attacks of different sharpnesses followed by a natural decay, creating a sound that hints at 'glass marimba meets music box'. Rubber beaters produce the softer tones while tapping with a hard beater creates a much sharper tone. However, the attack shape can be adjusted to make it softer. Some of the longer treatments, especially those in the Warp section, create a cascade of crystalline texture, some very pad-like and well suited to ambient/relaxation styles as well as haunting cinematic scores.

The effectiveness of this instrument depends very much on how you use it. Individual notes, if left exposed, decay and evolve in a very organic way, whereas playing anything too busy risks losing the ethereal character of the instrument unless you pick suitably short sounds, in which case you can use marimba or piano-style playing techniques. There's a useful range of tonalities available, and when you add in the sustained and warped sounds, you can conjure up anything from resonant, glassy hits to pads and drones, all with a wonderfully organic quality. *Paul White*

$99

www.spitfireaudio.com

Best Service

Horizon Leads By Sonuscore

Kontakt Instrument

★★★★

I reviewed Best Service's Kontakt-based Dark Horizon in the July 2022 issue of *SOS*. This was produced in collaboration with Sonuscore and utilised the latter's expertise in creating multi-layered performance engines (as seen in products such as The Orchestra, The Score, Elysion or EastWest's Orchestrator). This same team has now launched a new (and complementary) title; Horizon Leads. This features the same four-layer performance engine but is furnished with a different style of sounds. So, if you liked Dark Horizon, then Horizon Leads might also appeal.

As with the earlier title, these sounds are provided with two different levels of presets, with some 80 individual instruments (the library totals 2GB of sample content), and approximately 150 'themes' (global presets), each of which combines up to four of the individual sounds alongside suitable settings for the performance and effects engines. In terms of the individual sound presets, the emphasis is on synthesized sounds but mainly with an organic/acoustic feel. While the sounds themselves don't perhaps break any revolutionary new ground, there is lots of very usable content here and they could easily provide a hybrid synthesized/organic element that would work in similar musical/scoring contexts to more traditional orchestral sounds.

With a little dab of the built-in delay and reverb effects, there are some cool options for melodic parts based upon many of the individual sounds. However, Horizon Leads really comes into its own when the sounds are combined into a theme preset. These can also be accessed via a neat tag-based browser and offer categories of mono- and poly-style lead sounds plus some 'animated' options. The former provide conventional playable melodic sounds but, by combining the various source sounds, and with suitable use of the effects and modulation options, there is plenty of expressive character to be found. The animated presets make use of the engine's clever arpeggiator engine for one or more of the layers. These are great for rhythmic arpeggiated parts and most are set up with the mod wheel enabling some very usable sound modulation. In terms of moods, they span gentle ambient, mystery, tension and away into more high-tech or sci-fi territory. It's all done with an organic overtone, though, and things don't generally get overly aggressive.

Like Dark Horizon, Horizon Leads is probably aimed primarily at media composers working in modern drama, mystery, sci-fi, or even some nature projects. The sounds themselves are very usable and the Sonuscore engine — once mastered — provides plenty of creative potential. These Best Service/Sonuscore titles are turning into a cool little series. Given the relatively accessible price, Horizon Leads ought to appeal to media composers working at almost any level.

John Walden

$99

www.bestservice.com

Cradle

State Machine Slow Drift

Plug-in Instrument

★★★★★

State Machine Slow Drift's designers suggest that its focus is on ethereal textures, though in reality it offers up a wide selection of preset styles, much as a hardware synth might do. These include basses, lo-fi leads, soft pads, FM-style bells and incisive lead sounds that have a little more attitude. The GUI is reassuringly straightforward with all the 'instant gratification' controls on the Home page. Access to the Synth, MIDI FX and Audio FX pages is via tabs.

Each patch can be made up of two layered sounds, the balance being controlled by a large central knob. Samples can be swapped out from the main page and any MIDI FX or Audio FX combinations can be locked so as not to change when exploring new presets. Even the 'deep' access isn't at all intimidating so this is an instrument that lets you get results very quickly. Synth accesses the expected controls for sound source, filter, and separate envelopes for both filter and amplitude as well as LFO parameter modulation, and these can all be set independently for each of the two layers. While the LFO can modulate amplitude, pitch, filter and pan, it doesn't include a link to the balance control between the two layered sounds, which I would have found useful. Even an option to invert the LFO phase would have done the trick. It does however offer a vibrato fade-in control and a tempo sync option.

The MIDI FX section offers a scale quantise facility with a broad selection of scale types, a chord generator, and an arpeggiator of up to 16 steps with velocity control over each step. The Audio FX section comprises EQ, Distortion, Flanger, Delay and Reverb. Effect editing is available via the Audio FX tab to the left of the screen, but there's nothing scary in there. In the main there's a choice of effect variation with just three or four further rotary controls for adjustment.

Despite the name and the suggestion that this is a synth suited to ambient music, I see it more as a general purpose instrument, and as with most synth presets, only a handful will actually grab your attention. However, it doesn't take long to create a wide-ranging set of ambient or lo-fi pads and gentle leads. The chord generator is excellent for dance-style effects where you want to use all major, all minor or whatever chord type throughout the piece, while the scale generator quantises whatever you play to the selected key and scale type. The arpeggiator, used in conjunction with the chord generator, provides a 'one-finger' way to explore *Stranger Things*-type lines, and the audio effects, while fairly basic, do add to the complexity of the final result. Given its low cost and ease of use, Slow Drift has to be seen as excellent value, even though not all the sounds are slow or drifty. Still not sure? There's a free 14-day demo so try it for yourself. *Paul White*

$59

cradle.app

[Audio examples of this month's libraries are available at www.soundonsound.com.]

SOS SHOWCASE

David Carson (Regional Sales Associate): david.carson@soundonsound.com

PhantomFocus
Precision Monitoring Instruments • Carl Tatz Design USA
WORLD LEADER IN NEAR-FIELD MONITORING
Nothing Else... Comes Close...
Back In Stock
The Original

PhantomFocus Master Engineer Series eChair™
"The only chair ever nominated for a TEC Award with lumbar support unattainable from any other chair"
www.phantomfocus.com
Carl Tatz Design® LLC, Nashville, TN (615) 400-5479

Kludge Audio
506 Equalizer

THE ANALOGUE EQUALIZER FOR THE 96 KHZ AGE
http://www.kludgeaudio.com/500

With premium sound and aesthetics, our panels and bass traps are lab-tested and handcrafted with the professional in mind. Treat your music and studio with LA Sound Panels.
424.262.0622
info@lasoundpanels.com
www.LaSoundPanels.com

Piano
Organ
Synth
Note-for-Note
Sheet Music - MIDI Files
TRANSCRIPTIONS
Learn Your Favorite Keyboard Parts Exactly As They Were Recorded
Visit ManyMIDI.com
Free Sample
Money Back Guarantee

StudioBassist.com
World-Class Bass Tracks for your song! – by Martin Motnik
"You don't know your song is missing a live bass until you hear the difference!"
• On over 1,000 released songs
• Pro-quality studio and gear
• Major international label credits
• Live sessions via Zoom
• Member of *ACCEPT*
Recording credits with
- Gregg Bissonette
- Joe Satriani
- Hal Blaine
- Don Randi
- Michael Landau
- Frank Gambale
- Chad Wackerman
and many more

MODERN CLASSIC
FatKeys™ VCO
SOUND SEMICONDUCTOR
soundsemiconductor.com
SSI2130

Women's Audio Mission
Training over 1,200 women & girls every year in the only recording studio in the world run entirely by women
WWW.WOMENSAUDIOMISSION.ORG

DO IT YOUR WAY
"BUILD YOUR OWN" WEB PAGES
Full control for design, editing + more.
REDCO AUDIO www.redco.com 1-203-502-7600

PLACID AUDIO®
"A decade of use and still everyone's favorite mic to use."
–Jason Rubal
Veteran Engineer
COPPERPHONE™
ALSO FROM PLACID AUDIO®
RESONATOR SERIES™
CARBONPHONE™ & TONE BOX
COPPERPHONE MINI™
CARBONPHONE RU-80™
For more information or to hear sound samples visit:
placidaudio.com

Your SSL 9000 console
NOW !
in Pro Tools
TANGERINE AUTOMATION
Also available for
SSL 4,6,8 k
Flying Faders
GML – Focusrite
THD-LABS
thd-labs.com
Montréal, Canada
SSL, Pro Tools, Flying Faders, GML, Focusrite, are registered trademarks of their respective owners.

SOS WHY I LOVE...

THE SUZUKI OMNICHORD

BEN BROCKETT

The Omnichord is an unusual electronic instrument created by the Suzuki Musical Instrument Corporation in 1981. It features a small metal touch plate known as the 'Sonic Strings', and sound is created by holding down a chord button and sweeping one's finger across the plate. It has experienced a deserved renaissance in recent years, having been championed by Damon Albarn among others, and with its unique leftfield tones and engaging and unusual playing mechanics, it's not hard to see why it has become sought after by sonic experimentalists.

I was first introduced to it one morning in 2010. Before a team meeting at the substance misuse service where I'd recently started working, Debs from admin thrust a briefcase-sized parcel into my arms. I presumed it was something to do with my new employment, perhaps some kind of drug testing kit or training manual, but instead, when I removed the paper I found an old leatherette case with the word Omnichord emblazoned across it in gold. A small group gathered. "What is it?" they asked. Having no idea, I opened it and found what looked like a giant brown retro hearing aid nestled within. A strangely textured metal strip ran down it, with a selection of buttons and knobs scattered across the body. It became apparent that this was a musical instrument, but certainly not one I had seen the likes of before.

"How do you play it?" my colleagues asked. Having no idea, I plugged it in and was immediately assailed by the rattle of a tinny 200 bpm bossa nova. I deactivated the rhythm button and tried another, and was rewarded with a throaty chordal drone — I was liking this already. But what of this metal strip? Pleasingly bobbly to the touch but seemingly serving no purpose, it was only when I stroked it whilst holding down one of the chord buttons that the real magic happened. The whole room was instantly bathed in an otherworldly shimmering glissando that faded away on a tail of lo-fi reverb. This surely must be the music of the gods, I thought, my eyes lighting up as my finger swept up and down the strip, creating cascades of sound that somehow seemed to be inherently digital and organic at the same time. I became aware that some of my colleagues were now looking at me strangely so, not wanting to create too much of a stir so early on, I reluctantly turned it off and closed the lid.

I felt it unlikely that the Omnichord would be standard issue for East Sussex County Council workers and spent most of the day puzzling over where on Earth, or space, it could have come from? It turns out it was from a dear and resourceful friend from university who'd remembered my birthday but who didn't know my new address. I've never fully lost the sense of wonder I felt when I first heard the Omnichord in full song, and I hope I never will.

AD INDEX

To Advertise in *Sound On Sound* please contact: **usadsales@soundonsound.com**

Ableton ... 53	Baby Audio ... 47	Groove Synthesis 3rd Wave ... 4-5	Red Panda Lab ... 41
ADAM Audio ... 87	Burl Audio ... 31	Heritage Audio ... 81	Sound Devices ... 103
Allen & Heath ... 15	Cranborne Audio ... 13	IK Multimedia US ... 43	Soundtoys ... 111
Amphion ... 77	DPA Microphones ... 79	ILIO ... IFC	SPL USA ... 129
AMS Neve ... 39	Eventide ... 99	Josephson Engineering ... 47	Sweetwater ... 19
Antares ... OBC	Expressive E ... 21	Kenton Electronics ... 49	Telefunken ... 105
Antelope Audio ... IBC	FabFilter ... 69	Lindell Audio (RAD Distribution) ... 33	Tone Projects ... 65
API Audio ... 11	Focusrite ... 61	Oeksound ... 17	Toontrack Music ... 89
Apogee ... 35	Genelec ... 93	Orchestral Tools ... 29	Universal Audio ... 23
Arturia Software & Hardware ... 27	GIK Acoustics ... 63	PSI Audio ... 97	Wave Arts ... 107
Aston Microphones ... 25	Grace Design ... 9	Radial Engineering ... 57	

antelope audio

128-CHANNEL THUNDERBOLT/USB AUDIO INTERFACE WITH 64-BIT AFC™

ORION 32+

Gen 4

UNMATCHED PRECISION. UNWAVERING LEGACY. ATMOS-READY.

→ Acclaimed conversion: Up to 130dB dynamic range, powered by 64-bit Acoustically Focused Clocking technology.

→ Extensive connectivity: Thunderbolt™ and MADI connectivity, both accommodating up to 128 channels for large recording and mixing sessions, live sound, and broadcast facilities.

→ Atmos compatibility: Optional surround and immersive audio monitoring, supporting over 23 audio formats with precise calibration and seamless workflow.

→ Streamlined control: Redesigned software control panel offering convenient monitoring controls, powerful preset recall and customizable routing matrix.

Learn more

antelopeaudio.com

MADI 128
Up to 128 MADI Channels

AFC™
64-bit Clocking

Thunderbolt™ 3 & USB 2.0

Immersive Ready

AUTO-TUNE®

PRO

VERSION 11

THE PROFFESIONAL STANDARD FOR PITCH CORRECTION

Now with new 4-Part Harmony Player, the iconic Auto-Tune effect, ARA2 support, and improved Graph Mode for surgical vocal tuning.

AUTOTUNE.COM

SCAN ME